

Abstract

With the advent of 4G mobile networks service providers are looking to migrate voice services to Voice over Internet Protocol (VoIP) for cost and revenue generating advantages. However the move to VoIP alone is not sufficient to address carrier concern for revenue loss and need to generate new revenues. As wireline voice services migrate to IP they will become part of suite of real time multimedia-oriented communication services running on IP networks that share common client-server arrangement. Such services include Instant Messaging (IM), Push-to-Talk (PTT), NetMeeting and other 4G services. To evolution of VoIP supports ability to offer several new services through a new service delivery platform known as IMS defined by 3GPP group. This paper examines the IMS delivery Architecture.

Keywords: VoIP, PTT, SIP, CSCF, HSS, MGCF, SDP, CAMEL, LNP, TCAP, ANSI-41, ISUP.

Introduction

IP Multimedia Subsystem (IMS) standardized by telecommunications world is a new architecture for implementing IP based telephony and multimedia services for mobile users. It defines architecture that enables convergence of voice, video and data over IP based infrastructure. By implementing IMS service providers can provide Internet like services on their existing 3G/4G mobile network. IMS is not one network, but different networks that interoperate seamlessly thus holding the promise of seamless converged voice/data networks. IMS was originally designed for 3G mobile networks, but it has extended to become a access independent platform supporting CDMA 2000, Wi-Fi, Wi-Max, including broadband fixed line access for the service delivery. It provides seamless connectivity between mobile, public and private networks and devices. This new standard is a boon to the service providers as they can provide more value added services to their subscribers by integrating their core networks using this platform.

IMS Architecture

IMS services architecture is a unified architecture that supports wide range of services enabled by flexibility of Session Initiation Protocol (SIP). IMS architecture can support multiple application servers providing traditional telephony and non-telephony services such as instant messaging, push to talk, video streaming, multimedia messaging etc. The IMS

architecture is a collection of logical functions which can be divided into three layers.

1. Transport and Endpoint layer sometimes also referred as Access layer.
2. Session Control layer
3. Application Server layer

Transport and Endpoint Layer

The transport and endpoint layer initiates and terminates SIP signaling to setup sessions and provide bearer services such as conversion of voice from analog or digital formats to IP packets using Real time Transport Protocol (RTP). This layer provides use of media gateways for converting VoIP bearer streams to PSTN TDM format. The media server provides

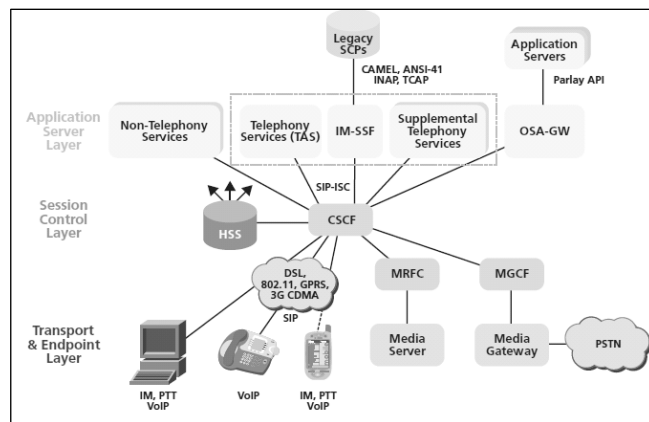


Fig1. Simplified view of IP Multimedia Subsystem

many media related services including conferencing, playing announcements, collecting in-band signaling tones, speech recognition, speech synthesis etc. The network elements at this layer are edge and core routers that provide all IP connectivity to core and access devices.



Fig.2 Endpoints and Transport layer elements

Session Control Layer

The session control layer contains Call Session Control Function (CSCF), which provides the registration of endpoints and routing of signaling messages to appropriate application server. The CSCF interworks with transport and endpoint layer to guarantee QoS across all services. The session control layer includes the Home Subscriber Server (HSS) database that maintains unique service profile for each end user. The end user’s service profile stores all of the user service information and preferences in a central location. This includes an end user’s current registration information (i.e., IP address), roaming information, telephony services (i.e., call forwarding information), instant messaging service information (i.e., buddies list), voice mail box options (i.e., greetings), etc. By centralizing this information, applications can share information to create unified personal directories, multi client type presence information and blended services. This centralized arrangement also greatly simplifies the administration of user data and insures consistent views of active subscribers across all services. The session layer also includes Media Gateway Control Function (MGCF), which interworks the SIP signaling with signaling used by the media gateway (H.248). The MGCF manages the distribution of sessions across multiple media gateways. The Media Server Function Control (MSFC) provides similar function for media servers.

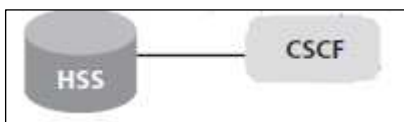


Fig.3 Session Control Layer elements

Application Server Layer

The application server layer contains the application servers, which provide the end-user service logic. The IMS architecture and SIP signaling is flexible enough to support a variety of telephony and non-

telephony application servers. For example, standards have been developed for telephony services based on SIP Session Initiation Protocol and SDP Session Description Protocol, and IM services based on SIP extensions for Instant Messaging. The application server is responsible for service specific logic i.e. Telephony Application Service (TAS), IP Multimedia Services Switching Function (IM-SSF), Supplementary Telephony Application Service (S-TAS), Non Telephony Application Service (N-TAS), Open Service Access Gateway (OSA-GW).

Telephony Application Service

The IMS architecture supports multiple application servers for telephony services. The Telephony Application Server (TAS) is a back-to-back SIP user agent that maintains the call state. The TAS contains the service logic which provides the basic call processing services including digit analysis, routing, call setup, call waiting, call forwarding, conferencing, etc. The TAS provides the service logic for invoking the media servers to support the appropriate call progress tones and announcements. If the calls are originating or terminating on the PSTN, the TAS provides the SIP signaling to the MGCF to instruct the media gateways to convert the PSTN TDM voice bit stream to an IP RTP stream and to direct it to the IP address of the corresponding IP phone.

As part of executing the telephony call model, the TAS provides the Advanced Intelligent Network (AIN) call trigger points. When a call progresses to a trigger point, the TAS suspends call processing and checks the subscriber profile to determine if additional services should be applied to the call at this time. The subscriber profile identifies which application servers should be invoked. The TAS formats a SIP IP Multimedia Service Control (ISC) message and passes call control to the appropriate application server. This mechanism can be used to invoke legacy AIN services or to invoke new SIP based applications servers.

A single IMS can contain multiple TASs that provide specific features to certain types of endpoints. For example, one TAS might provide the IP Centrex business features (i.e., private dialing plans, shared directory numbers, multiple call appearances, Automatic Call Distribution (ACD), attendant services, etc.). Another TAS might support PBXs and provide advanced Virtual Private Network (VPN) services. The multiple application servers can interwork using SIP-I signaling to complete calls between the different classes of endpoints.

IP Multimedia Service Switching Function (IM-SSF)

The IP Multimedia – Services Switching Function (IM-SSF) provides the interworking of the SIP message to the corresponding Customized Applications for Mobile Networks Enhanced Logic (CAMEL), ANSI-41, Intelligent Network Application Protocol (INAP) or Transaction Capabilities Application Part (TCAP) messages. This interworking allows the IP Phones supported by IMS to access services such as calling name services, 800 services, Local Number Portability (LNP) services, one number services, and more.

Supplementary Telephone Application Service (S-TAS)

The application server layer can also contain standalone independent servers that provide supplemental telephony services at the beginning of a call, at the end, or in the middle, via triggers. These services include click to dial, click to transfer, click to conference, voice mail services, IVR services, VoIP VPN services, prepaid billing services, and inbound/outbound call blocking services.

Non Telephony Application Service (N-TAS)

The application server layer can also contain SIP based application servers that operate outside of the telephony call model. These application servers can interwork with endpoint clients to provide services such as IM, PTT, or presence-enabled services. By implementing these non-telephony SIP based services in a common IMS architecture it is possible to interwork telephony and non-telephony services to create new blended communication services.

One example of such blended service is a converged click-to-contact buddy list that displays end user's presence and availability information, and provides a point and click interface across multiple communication services (telephony, IM and PTT). Another example is the use of a single pre-paid services account for telephony and VoD services.

Open Service Access Gateway (OSA-GW)

The IMS architecture allows service providers the flexibility to add services into their VoIP networks by interacting with legacy applications or by integrating SIP-based application servers that they purchase or develop themselves. In addition, service providers want to allow their customers to develop and implement services that leverage the VoIP network resources. For example, an enterprise may want to voice-enable or IM-enable some back office operations to automatically initiate a call or an IM if an order is about to be

delivered. This could be triggered by the location information of a wireless PDA carried by the delivery person. However, frequently the enterprise application developers have IT backgrounds and are not familiar with the variety of complex telephony signaling protocols (i.e., SS7, ANSI41, CAMEL, SIP, ISDN, etc.). To provide a simple API for communications services, the Parlay Forum, working closely with the 3GPP and ETSI standards development organizations, have jointly defined a Parlay API for telephony networks. The interworking between SIP and the Parlay API is provided in the Open Services Access – Gateway (OSAGW) that is part of the application server layer of the 3GPP IMS architecture. As described above, other applications servers provide the interworking between SIP and the telephony protocols (ANSI-41, CAMEL, INAP, TCAP, ISUP, etc.). The OSA-GW allows the enterprise-based Parlay applications to access presence and call state information, set up and tear down sessions, and to manipulate legs of a call. The OSA-GW implements the Parlay Framework, which allows the enterprise applications servers to register with the network and manage access to network resources.

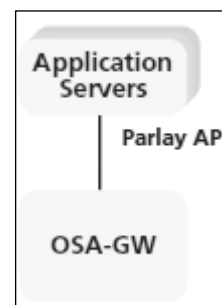


Fig.4 OSA-GW elements

Evolution of IMS Architecture

Most of the services described in the previous examples were broadband voice and data services. However SIP signaling and the IMS architecture can also support advanced broadband multimedia services. These services include Broadcast TV using multicast IP video streams, video-on-demand, video surveillance, video telephony, video conferencing, virtual classrooms, and more. These services can be implemented by equipping the network with additional multimedia application servers and endpoints.

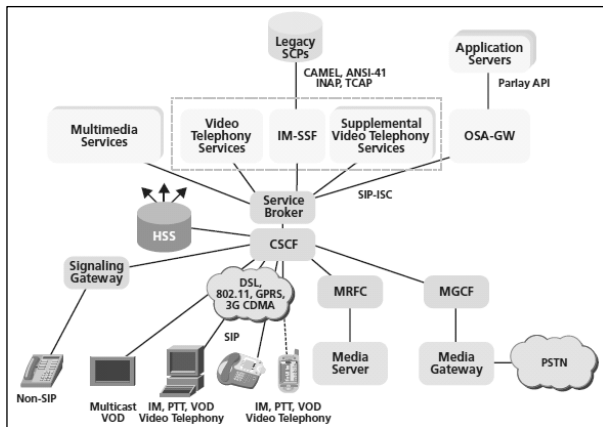


Fig. 5 Advanced IMS Services

As broadband multimedia services become more widely used, it will be necessary to move beyond the basic QoS mechanisms used today. In addition to monitoring the available bandwidth it must be possible to control the number of active real time communications sessions. With the IMS architecture, endpoints and applications servers for VoIP and broadband multimedia services send their session initiation requests through a common CSCF element. The CSCF can interact with the transport and endpoint layer of the network to assess current traffic levels and can deny requests for additional sessions.

Conclusion

IMS is an integral part of next generation telephony. IMS provides Multi service, Multi protocol, Multi access IP based network which is also secure, reliable and trusted. It is an opportunity for the service providers to migrate their existing legacy, fixed and mobile networks to IMS to enable new converged network that can not only provide legacy telephony services but also provide advanced real time and non real time services. Thus IMS is the both a challenge and opportunity for telephony, applications and services in coming decade.

References

- [1] 3G IP Multimedia Subsystem- Gonzalo Camarillo, Miquel Angel Garcia Martin, Wiley
- [2] IP Multimedia concepts and services- Miikka Poikselka, Wiley
- [3] IPMultimedia Subsystem Principles and Architecture- White Paper, Simon Zanty
- [4] Alcatel-Lucent IMS Solutions guide-Alcatel-Lucent
- [5] <http://www.3gpp.org/Technologies/Keywords-Acronyms/article/ims>