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Fixed Mobile Convergence – Ims Approach

By Ashwini Patil & H K Sawant
Bharati Vidyapeeth Deemed University College Of Engineering, Pune

Abstract - The paper is aimed at studying and analyzing the network performance parameters of SIP protocol. SIP is content based protocol, in which various message are required to be transacted so that a session could be created, terminated or modified. Therefore, the objective is to analyze various SIP activities and the delay incurred in session start-up under various network conditions. Proper functioning of IMS platform is dependent on optimum performance of several protocols specified in the standard. Nearly all of the protocols used in IMS are standardized by the IETF. Some of the major protocols are SIP, SDP- signaling protocol, DIAMETER- improvised version of RADIUS protocol,COPS- Common Open Policy Service, H.248- descendant of MEGACo, RTP/RTCP- Real Time Protocol/Real Time Control Protocol, etc. Out of all these, Session Initiation Protocol (SIP) is the prominent protocol used to create, terminate and modify the sessions initiated by the user. In order to improve the performance parameters, this is the area where most of the research work is centralized. Hence, to study various aspects of SIP protocol with respect to the network performance is of great interest.

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Strictly as per the compliance and regulations of:

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I. Introduction

In today's market scenario, it is seen that the mobile segment is growing with the faster speed all over the globe. Considering the Indian telecom sector, it is noticed that our telecom network is the second largest network in the world and it is the fastest growing network in the world. That's why Indian telecom sector is the true representative of the whole world. Following important issues are cropping up in the telecom sector and constraining the telecom growth.

a) Volume of telecom traffic is rising day by day and tariffs are falling down abysmally low.
b) Launching of high data speed services like 3G and BWA is on the anvil. Content based services and VAS are on the rise.
c) Fixed line network is reducing and competing with the wireless services. The fixed line broadband growth is not able to meet the market demand. There is a scarcity of spectrum and it is adding more expenditure towards erection of extra infrastructure/BTS's in a defined geographical area. Call drops are more due to multiple hand-offs in small area, which is degrading the Quality of Service (QoS).

Broadband on WLAN's is an integral part of the Fixed wireless Network, which can offer connectivity at outdoor as well as indoor, has not established the roots prominently in many countries till today. Competitive pricing and aggressive marketing are greatly affecting the churning of customers. The solution to all these problems is the Convergence of Fixed and Wireless services.

Author [α] : Department of Information Technology, Bharati Vidyapeeth Deemed University College Of Engineering, Pune-46
E-mail : hemant1960@gmail.com

a) Present Trends

Excluding the demarcations evolving between business and personal lives, new styles are emerging, which are demanding for services that are required at business and personal level. Now a days short message can be sent between mobile devices and fixed line phones. Service providers can provide seamless services over the common network infrastructure on wireless and wire-line broadband networks. The implementation of IP and SIP makes it possible to establish multiple sessions over an IP network or multiple networks and gives perception that there is a single network.

b) Convergence

With the technological advancements, the differences between fixed and wireless networks and hence the respective services are becoming unnoticeable. This convergence can allow a mobile call of a user to be delivered on s fixed phone or a fixed phone call can be delivered on a mobile device as per the suitability of the user. The main objective being that the mobile services and fixed services can be established on a single device which can switch between networks. Fixed Mobile Convergence can be achieved via user mobile device which can support wide area (cellular) access and local area (Wi-Fi) access. The concept of convergence is more user centric than network access technology.

As per the ITU-T recommendation Q.1761 on the principle and requirements for convergence of fixed and existing IMT-2000 system defines Fixed Mobile Convergence (FMC) as "A mechanism by which, on IMT-2000 user can have his basic voice as well as other services through a fixed network as per his subscription options and capability of the access technology".

c) Convergence Characteristics

Converged Customer Premises Equipment (CPE) – e.g. The services that a user is accessing through different gadgets can be made available on a single Wi-Fi phone. Thus the Wi-Fi phone can become the converged CPE. Convergence brings up the concept of personalization of services in fixed line services as these services are used at par with the wireless services. Irrespective of the configuration, the converged services bring out the unification of different

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aspects like freedom of mobility with the security, quality of service, higher bandwidth and lower costs of fixed-line services. It gives uniform communication experience at home and away from home with convenience and freedom of movement.

The converged network makes use of various radio spectrums and different technologies as a backbone infrastructure. The multi-radio infrastructure enables the cooperation of existing radio networks to combine their spectrum-efficient capabilities, whereby high-quality mobile multimedia services shall be provided.

II. Literature Survey

In the past few years, the evolution of cellular networks has reflected immense success and growth which was experienced by Internet in the last decade. This leads to networks where Internet Protocol connectivity is provided to mobile nodes. The result is third generation (3G) networks where IP services such as voice over IP (VoIP) and instant messaging (IM) are provided to mobile nodes (MN) in addition to connectivity. With import of Wi-Fi, Wi-MAX, digital video broadcasting, satellite, internet, etc. the current architecture of telecommunication network has been facing a challenge of interoperability.

Hence, the solution was to devise an interoperable platform which can provide the services irrespective of the access technology. The basic approaches to convergence are

UMA: Unlicensed Mobile Access (UMA) is a new technology that provides access to GSM services over Wireless LAN or Bluetooth. It also challenges the assumption of closed platform, since it is relatively easy to implement a UMA phone purely in software running on standard PC hardware and operating systems. In the UMA solution, exiting cellular network remains unmodified, and a new network element, the UMA Network Controller (UNC), is introduced. UNC acts as a gateway between the mobile operator core network and Internet or a broadband IP access network such as ADSL or cable. The phone connects to the IP network using a standard WLAN or Bluetooth access point. Since GSM/GPRS core security mechanisms; new mechanisms are defined only for protecting the communication between the phone and UNC.

SIP : The Session Initiation Protocol (SIP) is a signaling protocol used for establishing sessions in an IP network. A session could be a simple two-way telephone call or it could be a collaborative multi-media conference session. The ability to establish these sessions means that a host of innovative services become possible, such as voice-enriched e-commerce, web page click-to-dial, Instant Messaging with buddy lists, and IP Centrex services. SIP is a request-response protocol that closely resembles two other internet protocols, HTTP and SMTP; consequently, SIP sits comfortably alongside Internet application. Using SIP, telephony becomes another web application and integrates easily into other Internet services. SIP is a simple toolkit that service providers can use to build converged voice and multimedia services. SIP is an "application-layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants. These sessions include Internet telephone calls, multimedia distribution, and multimedia conferences."

IMS: IP Multimedia Subsystem (IMS) is referred as the heart of NGN. The 3GPP has published a number of specifications that define the IMS as the part of the 3G wireless environment which will "enable the convergence of, and access to, voice, video, messaging, data and web-based technologies for the wireless user." The 3GPP defined the IMS specifications in support of the Universal Mobile Telecommunication System (UMTS). The UMTS is the 3G evolution of the GSM.

IMS is the envisioned solution that will provide new multimedia rich communication services by mixing telecom and data on an access independent IP based architecture, defined in 3rd Generation Partnership Project (3GPP), 3rd Generation Partnership Project 2 (3GPP2) and Internet Engineering Task Force (IETF) standards.

The aim of IMS is to provide all the services, current and future, that the Internet provides with roaming facilities. To achieve these goals, IMS supports peer-to-peer IP communications between existing technology standards while providing a framework for inter-operability of voice and data services for both fixed (POTS, ISDN) and mobile users (802.11, GSM, CDMA, UMTS). It provides session control, connection control and an application services framework with both subscriber and services data, while allowing interoperability of these converged services between subscribers. IMS truly merges the Internet with the cellular world; it uses cellular technologies to provide ubiquitous access and Internet technologies to provide appealing services.

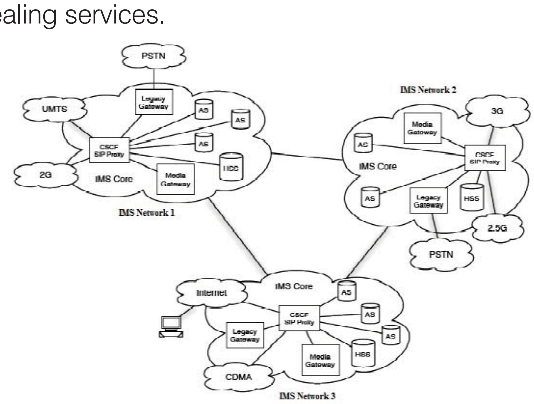

Figure. 1 IMS Overview : NGN Convergence

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III. Layered Architecture of IMS

The definition of IMS by 3GPP is an all packet core network that is able to accept all types of access networks i.e. WiMAX, CDMA, GSM in order to deliver a wide range a multimedia services which can be offered to the user by any device connected to the network. The use of SIP in IMS enable the support of IP-to-IP sessions over any wire-line connection system like DSL as well as wireless networks like Wi-Fi, GSM and CDMA. The IMS allows the inter-working between the traditional TDM networks and the IP networks.

a) Access Layer

The IMS architecture is independent of any access bearer. In the mobile networks the access layer could be any or a combination of the following: General Packet Radio Service (GPRS), Enhance Data Rate for GSM Evolution (EDGE), Code Division Multiple Access (CDMA), Wireless Interoperability for Microwave Access (WiMAX), Universal Mobile Telecommunication System (UMTS) and Wireless Local Area Networks (W-LAN or Wi-Fi). The fixed line networks makes use of Asymmetric Digital Subscriber Lines (ADSL) and cable network accesses.

b) Transport Layer

This is an all IP network which consist of IP Routers. These routers are Label Edge Routers and Core Switching Networks. The IP/MPLS (Multi-protocol cable Switching) is the transport layer technology for IMS platform. MPLS defines a mechanism for the forwarding of packets in a router network. Due to its flexibility, the MPLS has become the default IP transport network for the Next Generation Networks making use of the IMS core in order to reliability and Quality of Service.

c) Session Control Layer

This layer consists of network control servers which are used for the management of calls, establishment and modifications of sessions in the IMS platform. Two main elements in this layer are the Call Session Control Function (CSCF) and the Home Subscriber Server (HSS). These two elements form the core of the ISM architecture and are sometimes referred to as SIP Servers. The CSCF provide end-point registration and routing of SIP signaling messages and provide interworking with the transport layer for guaranteed QoS of all services. The HSS is the database that is used to store the subscriber service profiles and service triggers. The other information stored by the HSS includes the dynamic data of the subscribes like location information.

d) Application Layer

This layer makes use of application and content servers to provide value-added services. The Application Server (AS), the Multimedia Resource Function Controller (MRFC) and Multimedia Function Resource processor (MFRP) form the core of this layer. The AS is responsible for the execution of service-specific logic i.e. user interaction with subscribers and call flows. The MRFP is also known as IP media server is used to provide media processing for the application layer. The media server is used to enable to delivery of some non-telephony services like Push-to-Talk (PTT), speech enabled services, video services and other services like conferencing, prepaid and personalized call-back tones.

The control and application layers are access and transport independent and this is very useful in order to ensure that the user is able access the ISM services required from any access network and from any connection device.

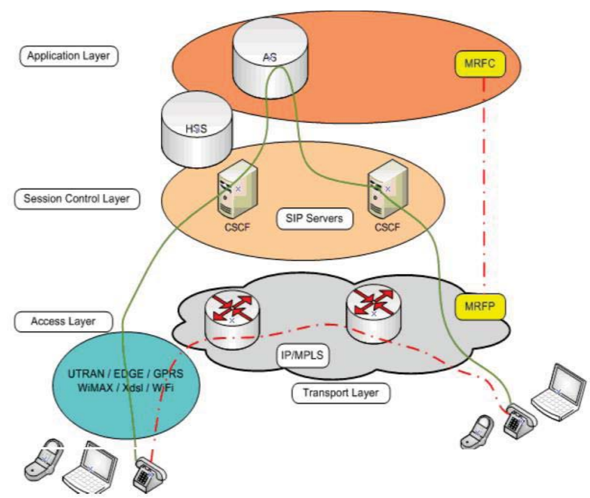

Figure 2 Layered Architecture of IMS

IV. IMS Architecture Design

In the General architecture of IMS, 3GPP standardize functions but not nodes. So we can say that the IMS architecture is a collection of functions linked by standardized interfaces. If any implementer want they can merge two functions into single node as well as they can divide single function into two or more nodes. An overview of the IMS architecture as standardized by 3GPP.[1,2] Here we include only most important nodes. The common nodes included in the IMS are as follows:

a) CSCF (Call/Session Control Function)

CSCF is a SIP (Session Initiation Protocol) server which processes SIP signaling in the IMS. CSCFs are dynamically associated, service-independent and standardized access points. It distributes incoming calls to the application services and handles initial subscriber authentication. There are three types of CSCFs depending on the functionality they provide.

P-CSCF (Proxy-CSCF): The P-CSCF is the first point of contact between the IMS terminal and the IMS network. All the requests initiated by the IMS terminal or destined to the IMS terminal traverse the P-CSCF. This node provides several functions related to security. The P-CSCF also generates charging information toward a charging collection node. An IMS usually includes a number of P-CSCFs for the sake of scalability and redundancy. Each P-CSCF serves a number of IMS terminals, depending on the capacity of the node.

I-CSCF (Interrogating-CSCF): The I-CSCF provides the functionality of a SIP proxy server. It also has an interface to the SLF (Subscriber Location Function) and HSS (Home Subscriber Server). This interface is based on the Diameter protocol (RFC 3588). I-CSCF retrieves user location information and routes the SIP request to the appropriate destination, typically an S-CSCF.

S-CSCF (Serving-CSCF): The S-CSCF is a SIP server that performs session control. It maintains a binding between the user location and the user's SIP address of record (also known as Public User Identity). Like the I-CSCF, the S-CSCF also implements a Diameter interface to the HSS.

b) SIP AS (Application Server)

The AS is a SIP entity that hosts and executes IP Multimedia Services based on SIP.

c) MGCF (Media Gateway Control Function)

MGCF implements a state machine that does protocol conversion and maps SIP to either ISUP (ISDN User part) over IP or BICC (Bearer Independent Call Control) over IP. The protocol used between the MGCF and the MGW is H.248 (ITU-T Recommendation H.248).

d) MGW (Media Gateway)

The MGW interfaces the media plane of the PSTN (Public Switched Telephone Network) or CS (Circuit Switched) network. On one side the MGW is able to send and receive IMS media over the Real-Time Protocol (RTP). On the other side the MGW uses one or more PCM (Pulse Code Modulation) time slots to connect to the CS network. Additionally, the MGW performs trans-coding when the IMS terminal does not support the codec used by the CS side.

e) HSS (Home Subscriber Server)

It contains all the user related subscription data required to handle multimedia sessions. These data include, among other items, location information, security information (including both authentication and authorization information), user profile information and the S-CSCF allocated to the user. The SLF (Subscription Location Function) is a simple database that maps users' addresses to HSSs. Both the HSS and the SLF implement the Diameter protocol.

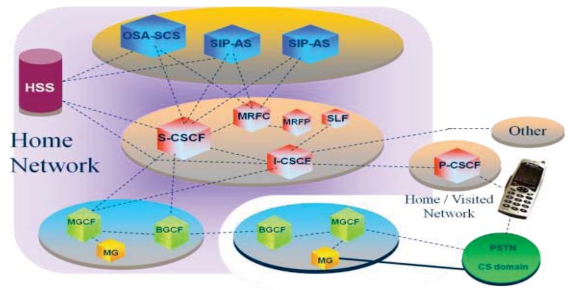

Figure 3 IMS Architecture

V. System Description

Several works analyzed the subject of SIP signaling from different angles. All these works did not associate any constraints to the access network and assumed an infinite bandwidth over the link between the User Agents, IMS entities and intermediate routers. This assumption is not true and the impact of SIP signaling on the network and its consequences on the SIP nodes must be evaluated. Apart from this, most of the studies were carried out with topology containing either fixed nodes or mobile nodes only; therefore the performance of a hybrid topology can be studied too.

So keeping in mind the drawback of the previous works, we propose a topology consisting of both mobile and fixed nodes with proper specification of suitable links between them.

Statistic to Be Obtained

1. Tunneled Traffic Sent/Received: Amount of traffic tunneled by Agents and de-tunneled by agents and hosts. It is given in packets/sec.
2. Jitter: If two consecutive packets leave the source node with time stamps t1 & t2 and are played back at the destination node at time t3 & t4, then: Jitter = (t4 - t3) - (t2 - t1) Negative jitter indicates that the time difference between the packets at the destination node was less than that at the source node. It is given in sec.
3. Packet End to End Delay: The total voice packet delay, called "mouth-to-ear" delay = network delay + encoding delay + decoding delay + compression delay + decompression delay. Specified in seconds.
4. Wireless LAN Delay: Represents the end to end delay of all the packets received by the wireless LAN MACs of all WLAN nodes in the network and forwarded to the higher layer. Specified in seconds.
5. Retransmission Attempts: Total number of retransmission attempts by all WLAN MACs in the network until either packet is successfully transmitted or it is discarded as a result of reaching short or long retry limit. Specified in packets.
6. Registration Traffic Sent/Received: Amount of Mobile IP registration packets sent/received. Mobile IP nodes and router register their addresses with

agent nodes (home/foreign) to receive mobile ip service.

7. Throughput: This statistic represents the average number of packets successfully received or transmitted by the receiver or transmitter channel per second.
8. Utilization: This statistic represents the percentage of occupancy of an available channel bandwidth with respect to time period, where a value of 100.0 would indicate full usage.
9. SIP Active Calls: Number of active calls at any given time.
10. Call Setup Time: Time to setup a call in seconds.
11. Calls Connected: It includes calls initiated by this node and also the incoming call requests.
12. Calls Initiated: Number of calls initiated at a particular SIP UAC node.
13. Call Duration: Duration of each call defined as the time at which the node got call connect confirmation to the time at which it got call disconnect confirmation.

VI. Conclusion

Next Generation Networks (NGNs) aims at providing a wide range of services to end-users over an access independent platform while allowing for better Quality of service (QoS), charging mechanism and integration of services as compared to conventional fixed or mobile networks. The NGN core network is known as IP Multimedia Subsystem (IMS) which is the Third Generation Partnership Project (3GPP) standardized core network for all IP-convergence of fixed and mobile networks.

The core functionality of the IMS is built on the Session Initiation Protocol (SIP), the Internet Engineering Task Force (IETF) standardized protocol for the creation, management and termination of multimedia sessions on the internet. Hence to study the signalling of SIP traffic is the subject of major interest.

In this work we have to analyze the impact of different types of traffic load with varying pattern of the SIP signalling carried over the network. We need to configure a hybrid network topology consisting of IMS entities I-CSCF, P-CSCF, S-CSCF, intermediate routers and SIP enabled fixed and mobile nodes. Then we have to create different scenarios and varied their respective parameters. Other signalling protocol like H.323, etc, do not support mobility but SIP handles this problem because of the Mobile IP support.

The IMS is based on Session Initiation Protocol (SIP) which is a text based protocol. The IMS will generally create additional signaling traffic in the IP based networks, so there is a need to take necessary precautions to minimize the signaling overload.

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