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Measurement Tools for IP Multimedia Subsystems (IMS)

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Abstract

In the field of telecommunication, to deliver high quality and low cost multimedia services to users is indeed a milestone. Next Generation Network (NGN) is an IP network which facilitates researcher to achieve this milestone.

IP Multimedia Sub-system (IMS) makes possible to deliver IP multimedia services to users. IMS uses Session Initiation Protocol (SIP) as a signalling protocol. Since IMS involves lot of signalling between IMS entities and other network elements, accordingly the signalling must guarantee quality of service. Test beds and measuring tools help us to perform various tests according to our need. These tests should be carried in early stages of development with designer in structured way according to well defined process and methods.

Initially this thesis aims to identify useful tools for IMS which can develop an appropriate IMS service creation environment. The basic task involves decoding of SIP message and to gather response time information. Finally to provide a proposal which tool is suitable for measuring signalling protocol regarding performance, exportability and hardware requirements

Acknowledgement

We dedicate the thesis to our beloved parents and respected teachers. We do believe that we would never ever able to perform any task of our life, if we don’t had the great help, guidelines and prayers of our parents and teachers. We believe that our parents and teachers are the blessings on us from our GOD, and it is our duty to respect them from our soul and always act upon their advices up to our last breath.

We are very thankful to supervisor and examiner of our thesis, that they guided us up the best they could and illuminate our brains from the knowledge of new era.

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Chapter 1

Introduction

Our new demands for chat and video sharing are the new challenges in communication culture. Now a day's operators really want such a business model which may support and ensure interoperable communication culture. In other words operators are looking for such a technique which can combine all benefits under one umbrella.

The IP multimedia subsystem (IMS), organized as the foundation for future wireless and wire line convergence, is the platform that will facilitate easy deployment on new, rich, personalized multimedia communication services that mix telecom and data services. In order to provide end user with innovative, cost-effective voice and data services the communication systems researchers, planner and designer will need to make full use of the technology **[1]**

Regarding industry concerns, we require network testing, network protocol analyzers and network monitoring systems, with some of the most advanced solutions on the market. We need such a sophisticated cellular testing solutions that supports GSM, GPRS, UMTS, CDMA2000 and future technologies. These systems should be capable of Voice Quality Management, which may allows service providers to deliver high-quality packet telephony services that back up their SLAs with assurance. These solutions should show the full array of drilldown and troubleshooting tools **[2]**

Internet Protocol Multimedia Subsystem (IMS) performs an appropriate development of the internet by giving quality and interoperability to telecom services. IMS also ensure security and better delivery of well known application like messaging, telephony, video, picture, text, voice etc. which guarantees profitable and successful telecom principles. IMS is a technology which allows the operators to retain its existing business model by addressing the key issues like enabler for convergence, fast & efficient service creation delivery, service interconnection and open standard **[3]**

In order to meet current and future monitoring demands of operators around the world, other requirements involve High online performance, Flexibility to support new services, features and capabilities, Scalability to handle higher traffic loads, more subscribers, transactions and services, Update and upgrade ease, High level of responsiveness to customer requests, Reasonable time to market and time in market, Balanced solution that avoids bottlenecks, Cost-effectiveness. **[2]**

For fast, efficient and reliable IMS services, Quality of Service (QoS) is very important and challenging. Since IMS involves lot of signaling, the precision at signaling/protocol level is worthy. For this, at first we need to extract the information (timestamps etc) at different layers from the protocol elements. The extracted information will be analyzed using different tools in order to observe the signaling/protocol behavior of IMS. Suggestions will be made in view of different experiment results in order to attain better Quality of Service for IMS signaling/protocol. **[3]**

The analysis part of this thesis work aims to identify a useful measuring tool for IMS signaling/protocol. This tool attached with IMS test bed will extract information from the protocol elements, which helps the researcher of MOBICOM Project to observe the behavior of IMS signaling/protocol.

The synthesis part of this thesis work will provide a proposal which tool to use for measurement based upon performance, cost, exportability and hardware requirement.

In Chapter 2, the keen aspects of Internet Multimedia Subsystem (IMS), architecture, main components and their functions are discussed as well as Session Initiation Protocol (SIP) is briefly discussed. In Chapter 3, different IMS test bed scenarios are clearly defined and their working is described. In Chapter 4, focuses deep concepts on IMS service creation environment and tool kits. In Ch 5, IMS service creation tools are defined along with different IMS traffic (signaling and data) analyzing tolls are identified which is the analysis part of thesis report. Chapter 6 proposed a IMS traffic (signaling and data) analyzing tool based upon performance, cost, exportability and hardware requirement.

Chapter 2

Technical background

2.1 Multimedia over internet

The internet is a global system of interconnected computer networks that interchange data by packet switching using the standardize internet protocol suit (TCP/IP). Internet is a unique system which interconnects to different autonomous computer networks. These computer network interchanges data by packet switching by using TCP/IP, which is a standardize internet protocol suit. Millions of networks, private and public are interconnected via internet using the medium (copper wires, fiber optic cables, wireless networks and several others). **[4]**

The main characteristics are online chat, file transfer, electronic mail, online gaming, and file sharing, different web resources, inter-linked hypertext documents instant messaging, voice over IP (VOIP), shared white board and video conferencing by using open protocols and much more ability to provide several new services.

The multimedia services can be delivered using TCP/IP through sources like Google and AOL but the question arises how these services can be managed, how they can be made secure, how to bill the services. Is TCP/IP is capable to overcome these challenges or we need to develop a new system, subsystem or transmission technologies to counter these challenges.

2.2 Migration from Legacy towards Convergence

By using the planned phases in spite of dismantling the entire network we can migrate from legacy network to all IP networks. There is a big difference between wireless and wire line network while migrating to all IP base network. The history of wire line operator starts from 1960, when they used plan to convert their network backbone to packet base rather then circuit switched, It was done because of addressing different issues like maintenance of different level of networks of different traffic types. The migration of wire line networks are more unordered then wireless network for example the technology chosen by wire line operators to meet their packet requirements is different as compared to wireless operator because wireless operator laid out other specific plan and strategies to meet their packet requirements. By having a look on 3GPP and specification of GSM evolution both are almost same when choosing the latest technology. In case of TDM base network, they migrated to packet base network in recent time. Asynchronous Transfer Mode (ATM) has been established as a new backbone technology when the BELL system companies migrate their ATM to packet switched network rather then circuit switched network. The migration of ATM towards IP base network made it more significant in the world networks. It is still cost prohibited when any one compare it towards the cost of IP. But still there are some lags that IP is unable to offer perfect and trust worthy services of the ATM transport network. **[5]**

In the case of TCP/IP the problem occurs for real time traffic such as voice and video. The induction of TCP was to transport data and more tolerance for packet loss and delays, but in case of real time traffic i.e. voice and video they are not as much tolerant due of this ATM was adopted quickly.

Answer for the primary need of all operators is the ATM protocol. We can also say QoS. In the Synchronous Optical Network (SONET) systems while using ATM on top of fiber backbone makes it perfect packet base backbone which has an ability to circulate and handle all their traffic. But the ATM protocol is not the best suitable protocol for voice and data because it delivers smaller packets to the end points much quickly. This is not suitable for data networks. The reason behind this while sending smaller packets, much more smaller packets are transferred to large data files which can create a big overhead within a network and make it non cooperative that's why Internet Engineering Task Force (IETF) concluded that TCP protocol is not sufficient and good enough to transport information for real time application. The IETF recommends a protocol namely Stream control transport Protocol (SCTP) which was the first protocol supports SIGTRAN. This protocol can be used for further expansion within the VOIP domains.

Above all this if an operator wants to take advantages of IP services they convert the backbone to packet base. For this purpose bandwidth and QoS should be keenly taken in to account are the key areas for the new era of telecommunication system.

The development of VOIP networks was to convert voice transmission in to a digital format from analog format. This new format transported over a new IP backbone. But the problem still persist because simply converting voice to packet format, it can be a problem during voice transmission because voice transmission is not tolerant of delays. This challenge is faced by the VOIP provider. Some has solved the quality of service and latency problem and peer to peer calls but still it miles away to overcome the actual laps.

The entire problem gives a birth to new though which can fulfill almost all the requirement of new telecommunication era according to 3GPP, which is IP multimedia subsystem

2.3 Internet Multimedia subsystems (IMS)

The IMS is well-known as an architectural framework which is used to deliver IP multimedia services **[4].** The main theme of IMS is to represent an appropriate path to serve internet services over GPRS then TISPAN updated this vision, 3GPP and 3GPP2 which translate an autonomous support for all network other then GPRS such as fixed LAN, wireless LAN and CDMA-2000. The origination of IMS is designed by 3GPP (3 Generation Partnership). According to 3GPP IMS uses session initiation protocol, which provides excess to various application regarding multimedia and voice from wireless to wire line terminal. An example is creation of fixed mobile conversions this works when the excess network is isolated by the horizontal control layer from the service layer. The common horizontal layer is a control layer and gives the provisions to any service, that they cannot have their own control function.

Nowadays subscribers are demanding more then current services like faster access network and service reliability. They require an appropriate and trouble free access, rich connected experiences, and security, personalization that they can control and share. In the running situation these subscriber are interconnected with both wired and wireless medium through wide spectrum but still the problem persist because many service provider failing to use their mobile infrastructure and broadband for the optimization of their revenue opportunities. The major drawbacks faced by service provider are guarantee in service performance, classification of application, provision of dynamic session, policy control and charge of multi IP services without costly infrastructure. The problem fir service provider is to manage and control subscriber base traffic like voice, video and data and billing for these applications accordingly. For this purpose they need a plate form that allows them easily to integrate and addition of services quickly and with quality. These are some challenges faced by many customer, business and technical development cooperation so that's why these subscribers are taking keen interest in IMS.

The development of IMS provides quality of services (QoS) which is required of enjoying different services voice, video, data and it also prevents from suffering poor services during real time multimedia services. It synchronizes quality of service provision recession establishment so the user can have predictable experience. Another merit of IMS is the charging of multimedia session appropriately. For example a user during his video conferencing session over a packet switched domain transfers a big amount of information mainly encoded audio and video data may experiences a large expenses due to generation of large amount of data these large expenses occur because the user using which operator services can not follow different business model to charge the user appropriately and the operator does not aware of the contents of those bytes transferred by user. Alternately if operator may know the actual service used by the user so the operation can use the alternative charging scheme which is more fruitful for the user. IMS resolve this problem.

IMS does not follow any particular business model. It is an open standard technology which allows operators to charge appropriately. When user uses some services the service information is invoked by the IMS which helps the operator to decide which flat rate should be used for this service whether time base charging, quality base charging or perform any other type of charging. The provision of the integrated services for the users is also the key reason for existence of IMS.The above explanation supports IMS, an answer for quality of services, integration of different services and charging.

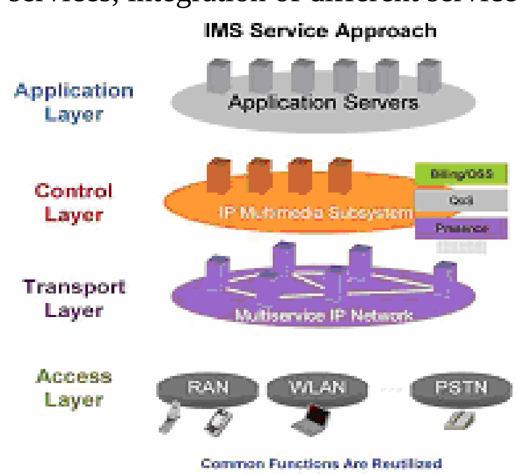

Fig: 2.1 IMS Service Approach [6]

The provision to user using different services in single and multiple synchronize session is the main ingredient of IMS. It is an application oriented concept which appeals to all types of service provider. The Fig2.1 shows IMS service approach model which separates the services and application layer services from transport layer entertaining greater flexibility.

2.4 IMS versus Non IMS Services

Many other technologies can provide internet services including real time multimedia services with acceptable QoS and price. Let say, two users establish video conferencing and sending multi media messages (MMS) to each other over a circuit switched domain in the same time they can check their email and surf web pages over the packet switched domain using GPRS. They can also excess the existence of more servers on the internet to find more people with whom they want to join the video conference. By using this technology we can provide good service and excellent QoS with no IMS at all. Then how IMS can take advantage on these technologies? **[5]**

It is because packet switched technologies IMS provide all services which responses more efficiently then circuit switched technology. The main differentiation of IMS from non IMS services is that it creates the service environment where every service can excess any session. The operators get the privilege to create richer service environment then it was before. For example the service can change the state of subscriber from busy to available or the service can display on the user screen that some person who is calling to user and his/her call is received or the same service can change the user status as busy and divert all the incoming calls to an email address instead then sending them to voice mail. Services in the network can easily access all the aspects of the session which can allow them to do many operations. Another important feature of IMS is that, IMS not depend on circuit switched domain so the inter working with different devices becomes easy.

2.5 IMS Architecture

IMS is a foundational network standard based on session initiation protocol. It is a network which allows a network provider to deploy audio, video and data services and these services seamlessly works across multiple wired and wireless networks. IMS is a collection of different function which is interlinked by the synchronized interfaces. The main flavor in IMS is its implementation which translate that the implementer is free to combines two function in one node or it can also split a single function in to two or more modes.**[5]** Similarly the node can also be represented by multiple times in a single network.

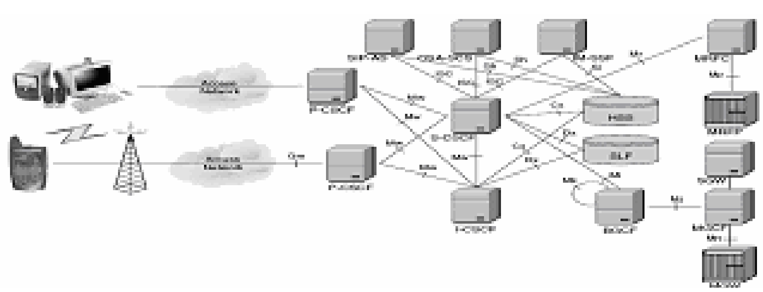

Fig: 2.2 3GPP IMS Architecture Overview [5]

2.5.1 Access Network

IMS network can be connected in various ways and all these ways used the standard Internet protocol. IMS networks in different places can be registered to several IMS terminals, and it doesn't effects if these terminals are roaming in different network or same network. The terminal requires Session Initiation Protocol (SIP) User agent (UE) and Ipv6 or Ipv4 sessions. These access networks can support mobile access, fixed access, wireless access and the other telephone system (POTS) non IMS compatible VOIP systems, H323 are supported through gateways.

2.5.2 Core Networks

a. Home Subscriber Server (HSS)

HSS is also known as user profile server function (UPSS). It supports the IMS network entities which handle calls. This is designated master user database and contain subscription related information, authorization, authentication and information of the user. It provides user physical location and behaves similar to Home location Register (HLR) and Authentication center (AUC).

b. Subscriber Location Function (SLF)

When multiple HSS's are used this functioned is used to map the user addresses. The diameter protocol is used for communication in between HSS and SLF. The diameter protocol is also known as AAA protocol i.e. authentication and authorization protocol.

c. Call/Session Control Function (CSCF)

CSCF is the presentation of several roles of session initiation protocol of server and proxies which are used to process SIP signaling packets in IMS.

I. Proxy Call Session Control Function (PCSCF)

First point of contact for IMS terminal is PCSCF which is also the SIP proxy. PCSCF can also be located in visited networks or in the home network. For the discovery of PCSCF terminal used Dynamic Host Configuration Protocol (DHCP) or program data processor (PDP) context. Session border controller is used by some other networks for the function of PCSCF.

During registration to an IMS terminal PCSCF is assigned and does not change during registration it inspect every messages and sits on the path of every signaling messages by using SigCom. SIP messages can be compress or de compress by PCSCF which reduces Round Trip time (RTT) over slow radio links. The user is authenticated and makes its association with IMS terminal by the establishment of IPSec security by the presence of PCSCF, which prevents from replay attacks and spoofing attacks and also assures the privacy of user because of this all other nodes does not authenticate user again and again and they trust the PCSCF. PCSCF includes the policy decision function and authorizes quality of services over media plane. The PDF may also be the separate function. Charging records are also generated by PCSCF.

II. Serving Call session Control Function (SCSCF)

The central node of signaling plane is known as serving call session control. It is also known as SIP server. It is located in the home network and perform the session control. The diameter interfaces Cx and Dx are used by the SCSCF to the HSS and the action translate downloading and uploading the user profiles. The necessary information is loaded from HSS as there is no local storage of the user profile in SCSCF. The SIP registration is properly handled and gives the privilege to bind the user location (IP and SIP addresses of the terminals are examples of user location). It inspects every message as it sniffs every path of all signaling messages. The forwarding of SIP messages to any application server is the decision of serving SCSCF for the betterment of services by using the electronic lookups it provide the better routing services and enforces network operator for exact policy. Due to several reasons of low distribution and high availability of the service, SCSCF can be multiple in the network. HSS assign SCSCF to user when interrogating call session control function (ICSCF) makes queries to HSS.

III. Interrogating Call Session Control Function (ICSCF)

At the edge of administrative domain another SIP function is located called ICSCF. By using NAPTR, SRV, types of DNS records in the DNS domain, IP addresses of ICSCF are published which helps in finding the remote services and also useful for forwarding points for the SIP packets to this domain. The interface Cx of diameter is used to retrieve the user location and function is done by HSS during ICSCF queries. In the second step Dx interface is used from ICSCF to SLF to locate the required HSS which routes the SIP request to designated SCSCF. In the release 7 and onwards ICSCF specification were changed as the entry point function is removed. And now this entry point function is a part of interconnection border control function (IBCF).

2.5.3 Application Server (AS)

Application server is connected with SCSCF through an interface using SIP and it executes services. It is also the host server and well developed for voice call continuity function according 3GPP. It operates in the SIP proxy mode, SIP back to back user agent mode, SIp user agent mode. It can be located in external third party network or the home network. The diameter Sh interface is used by the application server (AS), in case of its location at home network. This action gives provisions for HSS (home subscriber subsystem) queries. It has three components, which are:

- IP multimedia service switching function (IM-SSF)
- Open service access serviced capability server (OSA-SCS)
- Session initiation protocol- application Server (SIP-AS)

2.5.4 Media Server

Provides the media related function (playing tone announcements and mixing of voice stream). It is also known as Media Resource Function (MRF). There are two division of MRF one is media resource function controller (MRFC) and media resource function processor (MRFP)

2.5.5 Break out Gateways

The SIP server includes the functionality of routing based on telephones numbers and also well known as break out gateway control function (BGCF). When someone calls from IMS to circuit switched network i.e. Public Switched Telephone Network (PSTN) or Public land Mobile Network (PLMN) the BGCF is used.

2.5.6 PSTN Gateways

PSTN circuit switched network have interfaces with PSTN circuit switched gateways. Circuit switched network uses ISDN user part over message transfer part (MTP) for signaling purpose. On the other hand IMS uses SIP over IP. In the case of media services IMS uses real time transport protocol (RTP) while circuit switched network use pulse code modulation. PSTN gateways are the signaling gateways and have interfaces in between circuit switched networks and signaling planes. It transform stream control transmission protocol (SCTP) to message transfer part (MTP) a nd the purpose to pass ISDN user part from the Media Gateway Control Function (MGCF) to the circuit switched network. For the call control protocol conversion between SIP and ISDN user part, MGCF is used and have interfaces with signaling gateway (SGW) over SCTP. Across H.248 interface it controls the resources in the media gateway. The media gateway is an interface in between circuit switched network and media plane and it is done by RTP and PCM.

2.5.7 Media Resources

These are the components that operate on the media plane and IMS core function controls it. The IMS core function is MGW and media server (MS).

2.6 Types of Next Generation Network (NGN) Interconnection

2.6.1 Connectivity Oriented Interconnection (CoIx)

Irrespective of the levels of interoperability, it is the service provider and (physical & logical linked carrier) based on simple IP connectivity for example CoIx does not aware of end to end service when it is using IP interconnection so in the consequence service performance, QoS and security requirements are not assured however SoIx fully satisfied the requirements of NGN interoperability. There are two modes of NGN inter connection direct or indirect. Direct interconnection is connection between two domains and it has no any intermediate network domain and indirect interconnection at one layer defines the interconnection between two network layers with one or more intermediate network domains. The main function of intermediate network domain is the transit functionality of two other network domains. For the signaling and media traffic different interconnection modes may be used. **[4]**

2.6.2 Service Oriented Interconnection (SoIx)

It allows the carriers and service providers to offer their services over NGN platforms with complete control and signaling based session. It is also the physical and logical linking domain. It defines the complete levels interoperability and the levels if interoperability is dependent on quality of service, security and etc **[4]**

2.7 Charging

The phenomenon in which the users pay periodically for the services they are using is known as offline charging. On line charging, is also known as credit based charging. It is used for real time credit control of postpaid services or prepaid services.

In the offline charging all the SIP network entities involved in the session and they use DIAMETER Rf interface. The SIP entities are PCSCF, ICSCF, SCSCF, MGCF, MRFC, BGCF and AS. The interface send accounting information charging collector function (CCF) which is located in same domain. In formation is collected by CCF and makes call details record which is then sent to BS of the domain. IMS charging identifier (ICID) is a unique identifier carried by each session. The origination and termination of network are defined inter operator identifier (IOI). **[4]**

In online charging session charging function (SCF) is called by SCSCF. SCF is a regular SIP application server and can give signals to the SCSCF to terminate the session when the user is out of credit during the session. DIAMETER Ro interface is used by AS and MRFC for an event charging function (EFC) which is the function to deduct credits units from user account immediately. Immediate Event Charging (IEC) function does not authorizes the user if there is no enough credit units available. Event charging with unit reservation firstly reserve the number of credits in the user account and then give the authority to AS or MRFC and then consume credits units is deducted from the account.

2.8 Session Initiation Protocol (SIP)

For the creation and management of session, internet requires many applications. The session translates as an exchange of data between associations of different users. The applications used during the session among different users may involve some complication, when different users practices these application for example user may relocate between ends points, they may communicate in different medias, they may be addressed by multiple names. It is observed that for the real time multimedia session data like voice, video and text messages, a number of protocols have been tested and used.

So we talk about SIP. It is an application layer protocol that has an ability to make, terminate or modify multimedia session over the internet. Multimedia session is the data stream which flows from sender to receiver. The origination of SIP was developed within the SIP working group in the IETF and it is defined in RFC3261. Initially this protocol was designed to invite users in existing multimedia sessions and hyper text transport protocol & simple mail transfer protocol were used to make SIP. The main ingredient of SIP is, by some modification it can support all form of media, more secure and robust. In the other words we can say within an IMS network SIP protocol control everything that a user does. It enables internet end points by working concert with other protocols. These end points are known as user agents. SIP helps user agents to discover other user agents and make the session for them which they want to share. It enables the infrastructure of network host and these network hosts are called proxy server. User agents can send invitation to sessions, registration and other request to these proxy servers.

2.9 Session Description Protocol (SDP)

The important feature of SIP is session description protocol (SDP). SIP message contain session descriptions so the user can negotiate different parameters and media types for session establishment. The SDP describe that session being requested for example codec's of the voice call when it is being converted from voice transmission to digital transmission, should be identified in SDP. In this way the receiver will know how to decode the voice at its end. The message body of SIP request and SIP response carries SDP. There are three types of attributes defined for SDP session level description, time description and media description. **[5] [7]**

2.9.1 Terms of SDP

SDP is closely related to five terms namely conference, session, session announcement, session advertisement, session description. *Conference* is the set of two or more communicating users along with any specific software used by them. *Session* is following stream of data. This data contains the multimedia information of sender and receiver. S*ession Announcement* is the mechanism, when the session description is conveyed to several users like if the description was not explicitly by the user. *Session Advertisement* acts same as session announcement. *Session Description* is used for discovering and participating in the multimedia sessions. Session description conveys sufficient amount of information and also known as suitable format for this process. Series of attributes/value pairs translates a session. The single characters which are also known as attributes name, are followed by '=' and a value.

2.10 Architecture of SIP

There are four major components in SIP architecture. *SIP user agent* is the internet end point such as Convergence Bridge, personal computer, IP phone that is used to terminate, establish and modify session. The user agent (UA) can be act as user agent server or a user agent client. For the initiation of request a logical entry is used known as user agent client. For the generation of the response to a SIP request user gent server is used. *SIP redirect Server* is the server which generates the response to redirect the request at other location is a SIP redirect server and it is also known as user agent severs. *SIP proxy server* is an intermediate entity between different user agents for the routing of SIP messages to destination is the function of SIP proxy server. This proxy server acts the roles of both user agent server and user agent client. *SIP Register* processes SIP register requests. It is a user agent server that maintains mapping from SIP user names to addresses. It is the front end of the location service. **[5]**

2.11 Key Aspects of SIP

The key aspects of SIP are Naming and Addressing, Messages, Location Registration and Session Establishment and Termination.

2.11.1 Naming and Addressing

The identification of every SIP user is defined by SIP uniform resource identifier (URI), which behaves as an email address. The SIP URI has the host name and user name. SIPs URI is the secure SIP URI which defines encrypted and secure ways to transport SIP messages. Usually the secure SIP messages are translated by transport layer security (TLS). The SIP URI behaves as same as SIP URI but there is one difference which is the prefix SIPs other than SIP. **[5]**

The URI parameters defines, *Transport* is the parameter in which protocol is used to transport SIP messages. These protocols are, TLS, Simple Context Transport Protocol (SCTP), Transport control Protocol (TCP), User datagram protocol (UDP) or other protocol. *Multicast Address (Maddr)* is the simple form of loose source routing provided by maddr parameter. The parameter identifies the proxy which intents the SIP message to traverse in the direction of destination. The server which is specified by the maddr parameter parameter gets the request which override in the host field and directed to the server for example a URI with maddr =198.68.1.1 indicates that request is deliver to server with same IP address before the same request was sent to the address in the host field. *Time to Live (TTL)* is the multicast packet and time to live (TTL) value is defined by ttl parameter. It is used when the transport protocol is UDP and maddr is the multicast address for example if ttl is equal to ten then it can translate ten maximum numbers of hops, a multicast packet can travel. In *User* parameter *i*t can be a telephone number in the use info of the SIP URI or a user may have a name which looks like a telephone number.

The parameter user defines the difference between a real telephone number and user name which resembles telephone number. The *Method* of SIP request is identified by method parameter. It also indicates the primary function of the message for example the SIP request for registration is defined in method = REGISTER. *Lr* parameter makes the implementation of routing of SIP mechanism.

2.11.2 Messages

The SIP message can be a request message from client to server or can be a response message from server to client. RFC 2822 defines both request and response messages. The request message goes from UAC to UAS and response message goes from UAS to UAC. There are some methods which are involved in the SIP messages. *Invite* is used for the establishment of SIP session a user uses the message INVITE to invite other users. *Acknowledgement (ACK)* is used to confirm final response. *BYE* is used to terminate session. *Cancel* is used to cancel the SIP request. *Options* used to check the capability of server, it checks the server queries. *Register* is used for the registering the information. *Info* is used to carry ISUP and ISDN signaling messages. These are the information of session related control. *Subscribe* requests the state of dates and the current date from the remote node. *Notify* notifies the SIP node that the event has been occurred which is earlier requested by a SUBSCRIBE method. *Prack* is used for the provisional response Acknowledgment. In *Update*, the sessions are updated such as media stream and their codec's. *Message* is used to transfer instant messages (IM). *Refer* directs the user on the receiving side to other resources by using the information in refer request .The SIP messages are written in text format and it a text base protocol. The SIP message consist of the following a start line, one or more header fields, an empty line indicating the end of header field and an optional message body. There are six classes of status code in a SIP. *Provisional* translates that request has been processes after receiving. *Success* indicates that actions was understood, receive and accepted properly. *Redirection* is used to complete the request further action are required. *Client Error* translates that the server will not entertain this message because it contains bad syntax. *Server Error* is the error when server is failed to fulfill the request. In *Global Failure* **a**ny request cannot be entertain at any server. *Header Fields* are used to carry information for the routing of the request or for the managing of SIP session. The header field is composed of field name followed by a colon (:) and a field value. The fields are named as **TO**, **From**, **Subject**, **Call ID, Via, Contact, Maximum Forward (Max-Forward), Command Sequence (CSeq)**. **[5][7]**

2.11.3 Location Registration

SIP Register (special logical SIP UAS) is responsible for maintaining information current location, resides in user SIP service provider network **[5]**

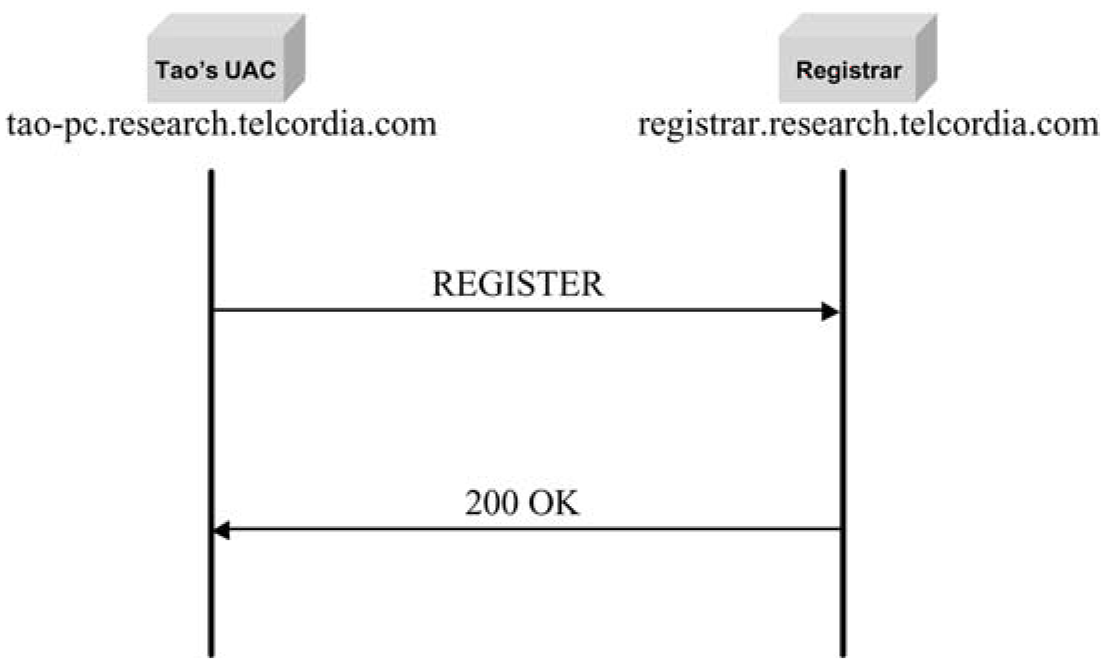

Fig: 2.3 Message Flow for SIP Registration [5]

The UA of user alpha sends the REGISTER message to registrar. If the address of registrar is not preconfigured the host part of address-of-record is used. Moreover the UA can also multicast the REGISTER to well known multicast address of all SIP- server. In

IPv4 the address 224.0.1.75 is a multicast address of all SIP servers. In IPv6 no multi cast address is assigned yet for this purpose.

After receiving REGISTER message, the request is process by registrar and decides whether the user is authorized to update the registration record. If the user fails the authentication and authorization process a 403 Forbidden will be returned. Once the user binding is updated successfully 200 OK messages is returned to the user.

2.11.4 Session Establishment and Termination

By using server mode or peer to peer mode, SIP session can be established. In case of peer to peer mode the call is directly established from a caller to a destination directly. In this case the caller does not use the path through any SIP server. It means that the caller knows the destination current location. In case of server mode the caller put a SIP request to a SIP server which is based in the destination SIP service provider network. The destination SIP server guides a caller to establish a SIP session then a SIP server forward the receive SIP request to the cal lee's destination on behalf of the caller. SIP proxy server is used for this purpose. The proxy server rewrites different parts of messages before forwarding it to the destination. On the other hand SIP server doesn't rely on the SIP request towards the destination. Other than this SIP server gives a response to a request with destination contact information and illuminate a caller, where it should connect further. For this case SIP server is a SIP redirect server. **[5]**

When the session initiates for example the session for audio, video or a game, the INVITE request is generated. The server establishes a session after the INVITE requests perform an action. The INVITE request is forwarded by proxies and it arrives one or more UAS. These UAS can potentially accept the invitation before they query the user. After queering the users these UAS's can accept the invitation and session establishes as these UAS's sends 2xx response. If they don't accept the invitation they send 3xx, 4xx, 5xx, 6xx responses depending upon the reason of rejection. The UAS before sending the final response can send a provisional response 1xx and give advice to UAC to contact the call user. After UAC receives one or more provisional responses it will get one or more 2xx or non-2xx final response. When UAC receives the final response it sends ACK for every final response it has received.

For the session termination we should understand the state of dialogue and state of session which are closely related. UAS creates a dialogue when the session is initiated with an INVITE request, so we have a response 1xx or 2xx and when the response complete the answer exchange it creates the session. If the INVITE request generates a non-2xx final response initially it terminates all session and all dialogue that were generated in the response of that request. By generating the request non 2xx it also prevents to create more session in the result of any INVITE request. For this purpose BYE REQUEST is used to terminate the session. When the dialogue received BYE REQUEST, the session associated with that dialogue is terminated. The callers UA sent a BYE REQUEST on early or confirmed dialogues and destination UA send a BYE REQUEST on confirm dialogue and does not send a BYE REQUEST to early dialogues. The calling parties UA must not sent a BYE REQUEST on a confirmed dialogue unless and until it receive acknowledgement of its 2xx response. **[5]**

Chapter 3

Test Scenarios of IMS

3.1 Multi service forum (MSF)

IP Multimedia Subsystem (IMS) is such kind of framework that claims important news and messages for next generation networks (NGN). It also forecasts the number of advanced services that converged networks will deliver in future. The core capabilities of IMS are consistent services without performance penalties, across multiple networks (unrestricted roaming). Unrestricted roaming facilitates users and devices that they can easily use and access the network anywhere in the system. So by this way we can eliminate the boundaries in between wireless, fixed & mobile networks. In simply, IMS is the name for true multimedia services that insures QOS to the user, weather they are using any type of value added services. **[8]**

Multi service forum (MSF) is an architecture that focuses a full peer network as complete IMS implementation. It demonstrates the interoperable environment, and the demonstrations are:

- By using Session Boarder Controller (SBC) and Bandwidth Manager (BM), MSF performs effective QOS and also enforces it.
- In the fixed mobile convergence environment (FMC), it gives the security interoperability.
- IP carriers interconnect/interworking.
- Between PSTN and NG N it applies interworking of priority calling.
- Service brokering and third party applications.

MSF (multi service forum), is well known global association of test equipment vendors, users, service providers and system suppliers, who are promoting and developing open architecture and multi-service next generation networks (NGN). MSF encourages the interoperability between different network elements and international standards. Its architecture and interoperable framework guarantees to combine next generation services and legacy in a one-unit network.

3.2 Scenario Overview

Before going into the test scenarios technicalities and functioning, we have to understand the network management system (NMS) of theses scenarios. Network management system (NMS) gives the provisioning and testing of MPLS/VPN network environment and verifies the automatic calculations for router configurations for MPLS/VPN and label switched paths (LSP). After that it confirms the configurations and gives provisions, so that theses configurations can be set up, deleted or modified in the multi vendor environment. The calculations for the network topology information, which are based on the full network traffic view and individual router configurations, are then generated. The initial setup of the network was conducted and tested. To check the routing connectivity, simple pings are established and sessions like Diameter, H248, and MGCP are performed. And at the end inter and intra domain call routing was performed.

3.3 Test Scenarios

3.3.1 Scenario 1 (Basic Call in Single IMS Domain)

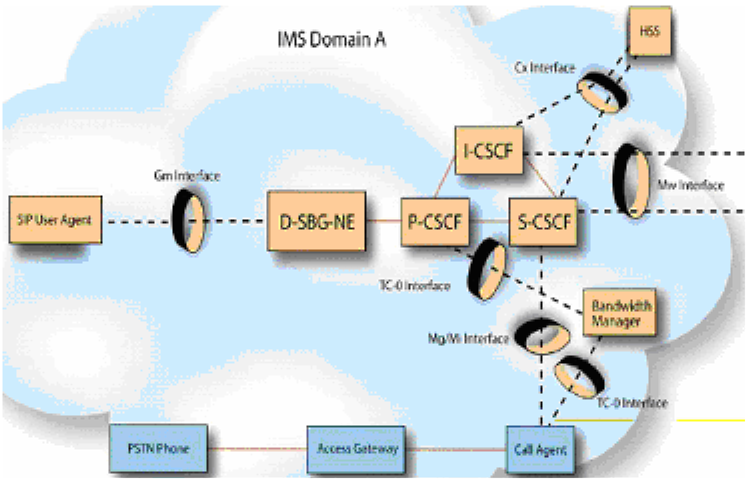

Fig: 3.1 Single IMS Domain [8]

CA: Call Agent
AGW: Access Gateway
HSS: Home Subscriber Server.
SIP: Session initiation protocol.
P-CSCF: Proxy call session control function.
S-CSCF: Serving call session control function.
I-CSCF: Interrogating call session control function.
BM: Bandwidth Manager (Resource and Admission Control)
S-SBG-NE: Signaling component-signaling session boarder gateway-network edge.
D-SBG-NE: Data/media component-signaling session boarder gateway-network edge.

The scenario in Fig 3.1 shows the basic building block of single IMS domain. The scenario 01 shows the various locations in the network from where IMS device can register. This validates the ability of IMS to have an access to both an IMS and non-IMS services. Figure 3.1 defines the MSF architecture in which non-IMS devices like PSTN can coordinate IMS users by establishing a session using call agent and access gateway. By using the SIP Mg/Mj interfaces, call agent accesses the S-CSCF. The calls from PSTN use the Mg interface through call agent into the IMS network. And Mj interface reverses the process of Mg interface. **[8]**

The call signaling between P-CSCF and UA is performed by SIP Gm interface. This interface also manages the calls into the IMS network. HSS (home subscriber function) has the information and profile of the user. For the appropriate call routing, Cx interface is used in between I-CSCF/S-CSCF and HSS. For the exchange of information between call session control functions (CSCFs), Mw interface is used, which acts like network-to-network interface (NNI) and directs the call to exact S-CSCF, if its domain is at different location. **[8]**

a. Scenario 1.1 (Mc Interface)

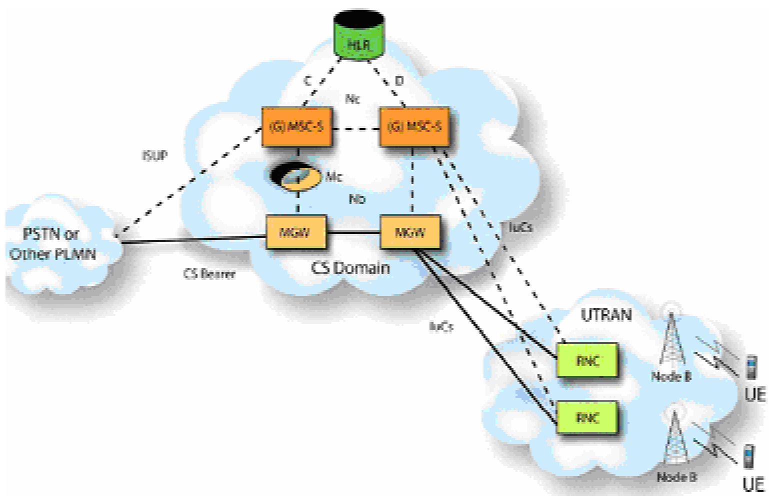

Fig: 3.2 Mc Interface [8]

UE: User Equipment
CS: Circuit Switched
MGW: Media Gateway
Node B: Node B (simulated)
HLR: Home Location Register
RNC: Radio Network Controller
PLMN: Public Land Mobile Network
MSC-S: Mobile Switching Center-Server
UTRAN: Universal Terrestrial radio access network

The Mc interface is communicating in-between MGW and MSC-S. This scenario validates (3GPP to 3GPP release 4 bearer independent core network) Mc interface between MGW and MSC-S. MGWs, independent scaling of MSC-Ss, inter-MGW voice transport are enabled by the implementation of BICN throughout the network, which gives batter voice quality, bandwidth utilization and transcoder free operation. Mc interface verifies handovers, general procedures, call clearing, establishment of basic call and so on. Indeed Mc interface gives the demonstration of the interoperability. There are some other interfaces, which also taken into account while verifying Mc interface. These interfaces are (Nc interface, in-between MSC-S & MSC-S) and (Nb interface, in-between MGW & MGW).

3.3.2 Scenario 2 (Single IMS Domain with Value Added Services (VAS))

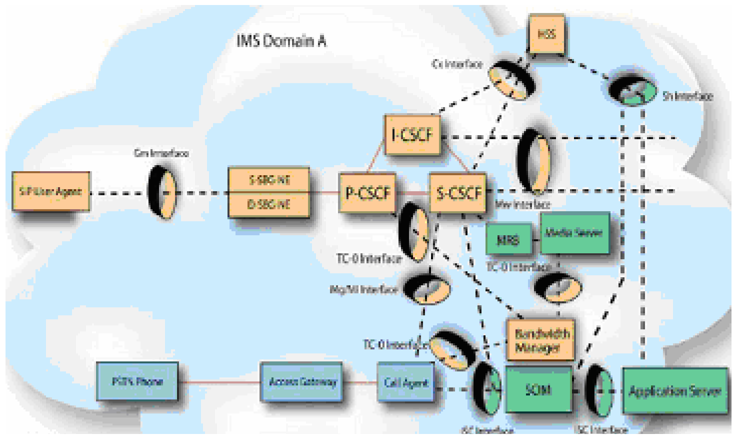

Fig: 3.3 Single IMS domain with VAS [8]

Media server: Media Server
MRB: Media resource broker
SCIM: Service Capability Interaction Manager
Application server: Parlay Gateway / Application
D-SBG-NE: Network Edge Session Boarder Controller-Data/Media Component
S-SBG-NE: Network Edge Session Boarder Controller-Signaling Component

The user can access the value added services by the validation of this scenario. These value added services includes the sessions of priority video and priority voice by using the third party applications and caller ID information. So in this way users can get registered from different points in the domain, and this capability and function are supported by some more interfaces shown in Fig 3.3.

The main significance of service capability interaction manager (SCIM) is that, it can pull services together from individual components. And these components may be placed on different platforms in the network. By using the interfaces Sh (SCIM) and Cx (S-CSCF), the service profile of the user can be retrieved from HSS. Static service profiles are handled by S-CSCF and dynamic & context dependent service profiles are handled by SCIM.

S-CSCF is enabled by ISC interface, which gives the path way to application servers. Theses application servers may further connect to media servers. For the optimistic allocation of media servers to sessions, media resource broker functions are employed. For the interworking between IMS domain and PSTN, media/signaling gateway is used in MSF release (3) architecture. The purpose is to support the priority calling, when the call originates in any network. For the triggering of call admission control (CAC), SIP resource priority header is used which means that priority calls are easily accessed to IMS network then non-priority calls, when the network is congested.

3.3.3 Scenario 3 (Interconnectivity between Two IMS Domains)

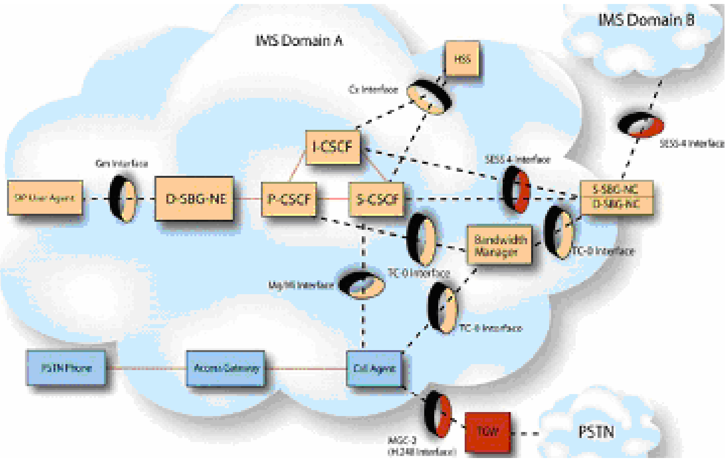

Fig: 3.4 Interconnectivity of two IMS domain [8]

TWG: Trunking Gateway
BGCF: Breakout Gateway Control Function
S-SBG-NC: Signaling Session Border Gateway-Network Core

This scenario gives the information of IMS interconnection between IMS domain and the users. It also validates the addition of second IMS domain in the same scenario, which translates as IMS as peer network. The scenario involves testing of NNI and S-SBG-NC (Signaling Session Boarder Gateway-network core), which are in-between the two IMS domains. Signaling Session Boarder Gateway-network core (S-SBG-NC) acts as a point of contact in between the two IMS domains. S-SBG-NC also performs the functions like negotiation, security relationships between S-SBG-NC's, hiding where required and controlling the network address translation NAT/firewall functions in their own domain. S-SBG-NC serves as application level gateway and also manages QOS.

To establish the IMS call signaling from the CSC, to the CSC and in-between IMS network, S-SBG-NC uses the interface namely MSF SESS 4. When the call is processed for the PSTN network and passed by the S-CSCF to call agent (CA), call agent (CA) uses the BGCF (Breakout Gateway Control Function), and the call is routed to the circuit switched telephone network. For the creating of media path between IMS domain and PSTN, Trunking gateway (TWG) is used. The scenario 3 gives the range of the network and its interconnectivity options. For the proper test, we have to make the basic call and media path should be established in each direction.

3.3.4 Scenario 4 (Roaming)

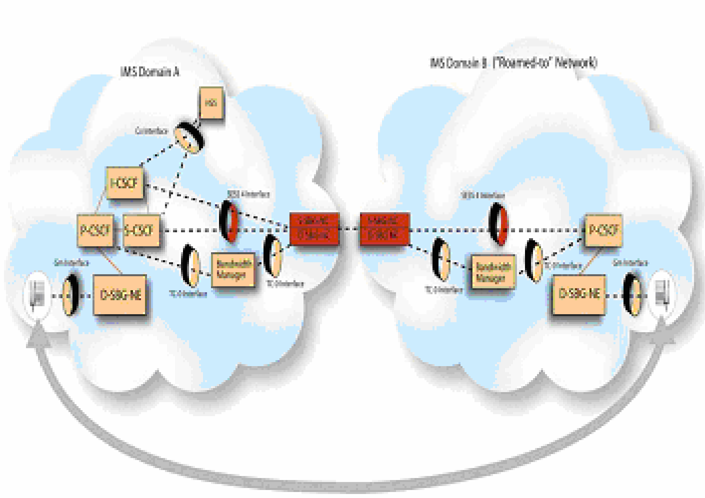

Fig: 3.5 Roaming [8]

This scenario gives the presentation of the roaming between two domains of IMS, and describes that how proper roaming is achieved. Signaling and media of both domains are tunneled back by the GGSN model in the subscriber's home network, which enhances the efficiency and serves as extended access network. P-CSCF (Proxy Call Session Control Function) model gives the privilege to the user to visit other network and S-CSCF (serving call session control function) gives the privilege to the user to connect back to its own home network. This scenario tests P-CSCF, because P-CSCF enables the end-to-end QOS and gives the optimization for media routing.

The domains of the visited and home networks are separate, which represents that the D-SBG—NC and S-SBG-NC pairs are deployed separately one on the edge of per network (on network to network interfaces). In between P-CSCF and Mw (I/S-CSCF) interfaces, there is an interface exists, which is SIP interface and it also passes through the pairs of SBGs. The scenario gives us the description of potential and appropriation of routing between end points, and maintains end-to-end QOS. This capability can enhance the system performance in future.

3.3.5 Scenario 5 (Roaming With Value Added Services)

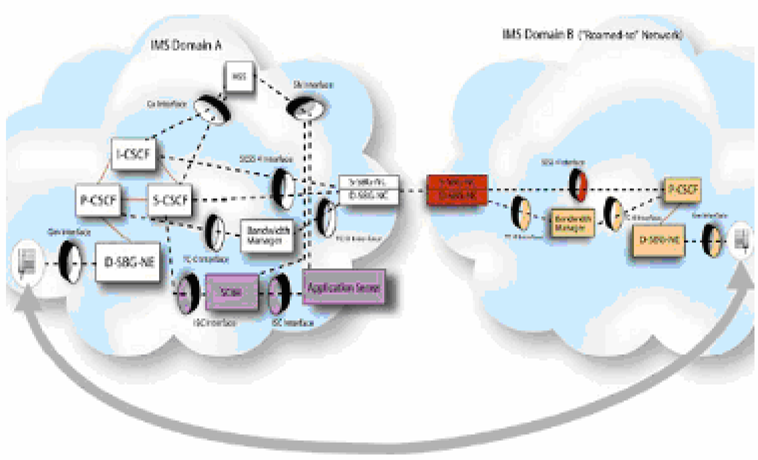

Fig: 3.6 Roaming with VAS [8]

This is the final and the fifth scenario, and gives the representation of the value added services described in scenario 02. It describes the roaming of the user in different network. It describes an ability of subscriber, when it accesses its services while roaming in different network. While the user is roamed in different network, its session is controlled by the S-CSCF of home network of the user. It is done by using application interface (ISC) that gives information to the application entities on the current access network.

Chapter 4

IMS Service Creation Environment & Toolkits

4.1 IMS Service Development

The main characteristics of IP convergence and IMS architecture show ability for the development and deployment of new services. According to business point of view, a businessman establishes their goals and priorities. In the full cycle of the service development, software development personals manage new services developments, and operations team takes care of running and deployment of these services. There are some steps of full cycle service development. These steps involve *Model the Service* which is the simulation of alternative scenarios, capturing flows as well as service activities. *Analyze Requirements* defines the requirements of the service, by visualizing different flows and user interaction models. It also captures the requirements for business and technology. *Design & Construct* are for the creation of high quality services; the team follows a development paradigm, and also translates the requirements for batter service. *Test* conducted for the acceptable performance. This work is done by quality assurance of the software. *Deployment* services are appropriate deployed to get complete coordination and management for best environment in service execution. In *Monitor* the feedback is taken by the comparison in-between actual values and projected values. After that important adjustments are made for batter business. **[9]**

4.2 Unified Service Creation Environment (USCE)

To support the full cycle service development, we need such kind of service environment, which is robust and has strategic importance. USCE has vast variety of set of tools, which offers best professional develops business driven services. These tools support different functions during service creation and roles in all software development life cycles. There are some areas of software development life cycles, which are mapped by these tools. These areas are defined as *Requirements & Analysis* which are Integration tools for data modeling use case development, business modeling, and requirement management. *Design & Construction* tools for run time analysis activities, architecture and design modeling, model driven development, component testing. *Software Quality* defines functionality; reliability and performance are the three dimensions of software quality. And these tools address theses dimensions. *Software Configuration Management* defines any change including version control is simplified and managed. Solutions are available for change and defect tracking, software asset management. *Process and Portfolio Management* is the integrated solutions, which can implement proven development process, assess and report progress and help the teams to manage the change and requirements. **[9]**

4.3 USCE (Unified Service Creation Environment) Toolkits

For the development of SIP and IMS applications, USCE includes some toolkits. These toolkits create SIP and IMS applications. The toolkits are named as, application server tool kit, IMS enablement toolkit and web services toolkit. In service creation environment, SIP and SIP/HTTP converged applications are created by application server tool kit, IMS foundation applications are created by IMS enablement toolkit and composite IMS applications are created by web services toolkit. **[9]**

In runtime environment, converged container for SIP and HTTP servlets are supported by application server through SIP during runtime process. Business process security, transactional integrity, consistency is executed by process server, which is high performance business engine. Service continuity, service oriented integration, java message service (JMS) is provided by enterprise service bus.

4.4 Application Server Toolkit

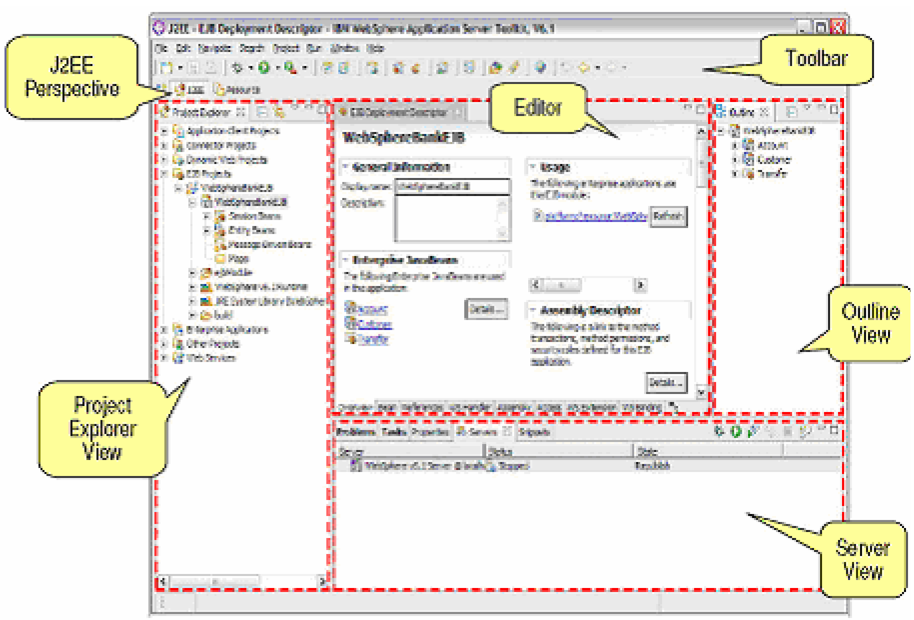

Fig: 4.1 Application Server Toolkit (AST) Workbench [9]

This toolkit is used to deploy, test and create applications that run within the application server. Application server toolkit is eclipse based integrated application development environment. The tools used in application server toolkit are integrated into a workbench, which is based on web tools platform and eclipse technology. This toolkit focuses from the application deployment tool to the application development tool and has some features. **[9]**

Tools for jython development
Web tools platform version V 1.0.2 is included
Tools support for JSR 168 port lets development
For the testing and publishing applications, tools are more improved.
Tools supporting JSR 116 SIP servlets application development

The application server toolkit workbench toolkit includes J2EE application, which can allow the creation of web services from different files like enterprise java beans, java beans, WSDL. By this characteristic we can create, deploy and modify enterprise java beans, container managed persistence (CMP) can be do-bottom up, back-end database can be supported for enterprise java bean (EJB) deployment code. Recently it was not possible and we could only do deploying and assembling of enterprise java bean (EJB), and could only modify their deployment descriptor.

4.4.1 Perspective of AST V6.1 Tools

The SIP application development tools supported by application server toolkit version 6.1 are translated through J2EE perspective. *Converged SIP/HTTP and SIP projects* are two projects namely Converged SIP/HTTP and SIP applications are represented by application server toolkit. *SIP Servlet Development (JSR 16)* **a**pplication server tool kit version 6.1 indorses the capability for the development of SIP servlet, which is based on JSR 116 specification. This is an addition to HTTP servlet development. *Import/ Export of SAR packages* defines that, from inside application server toolkit, Sip applications cannot be directly installed on application server like other J2EE or Web applications. So for this purpose we have to use import/export tool to export SIP application. It will be done to export SIP application as a part of enterprise archive (EAR) or standalone SAR package. Then we use the administrative console to install the package to the application server. *SIP Deployment Descriptor Editor* defined for the packaging and configuring SIP applications, editor is used. SIP application resource (SAR) is a new construct in which SIP applications are packaged. Http servlets and SIP servlets are encapsulated by these editors and are fully supported by application server toolkit version 6.1. In the application of the servlet, different parameters can be specified and configured by using SIP Deployment Descriptor Editor. For example if we want to add servlet mappings by using SIP Deployment Descriptor Editor, SIP messages routes to the exact servlet for processing. **[9]**

Some other tool features for application server version 6.1 are available.

- Server Tools used for the support of unit testing and debugging.
- They support SIP and Jython tools for application server specific-extensions
- Graphical editors are available for deployment descriptors and application server property files.

4.4.2 SIP Servlet Application Development

JSR 116 servlets, which are also known as siplets are developed by the application server tool kit. In this way we can develop and test SIP applications in the same manner we test and deploy J2EE applications. The group of source files, servlets, and resources, which are also known as session initiation protocol (SIP) applications can be managed in single unit. By using the application server toolkit, we can develop SIP only for HTTP and converged SIP servlets. **[9]**

a. SIP only Applications

For the development of new SIP application, we need a new SIP project wizard. The wizard creates SIP project and has additional information in the project menu like, SIP servlet, SIP project and converged project. SIP servlet give the application functions and gives the definition of servlet class name, super class, package and location. In the unique SIP mapping section, we use the deployment descriptor information to enter the SIP servlet deployment descriptor information. The wizards also define the interfaces and different methods. In the servlet class, stubs are created and we implement the servlet by developing code for stubs.

b. Converged SIP/HTTP Applications

The creation of hypertext transfer protocol (HTTP) and session initiation protocol (SIP) applications is supported by application server toolkit. The dynamic web project which is also known as converged project with SIP contents and also have web deployment descriptor files and SIP deployment descriptor files. If the web project already has HTTP servlets according to SIP specifications, it can make easy to check the symmetry between web.xml and sip.xml. SIP servlets can be added from the deployment descriptor editor, when the SIP deployment descriptor is created.

c. SIP Servlet Deployment

The SIP application resource (SAR) file is a deployable SIP application which is a basic java archive with ".sar" file extension. It also includes SIP deployment descriptor file. The extensions of web archive resource (WAR) export and import wizards, are SIP application resource (SAR) export and import wizards. The SIP project exporting is allowed by the SAR (SIP application resource) export wizard and tells us if we can include export files or not. During export process, SAR export operation has additional validations, which ensures the compliance with SIP specifications. This is the specific case of converged SIP/HTTP application. Servlet context manages the initialization parameters for SIP and HTTP, while the WAR file may have the converged application. The reverse of import wizard is the export wizard, while in import wizard we can import previously exported SAR file in our workspace. The creation of SIP project occurs when SAR file is a SIP only application. Otherwise SAR file is a converged SIP/HTTP application and then project with contents of SIP and HTTP will be created.

The SIP deployment descriptor (DD) tells us about the deployment of SIP applications. For the syntax of the deployment descriptor file, XML is used. For the SIP applications, information is stored in ".xml" file and for HTTP applications, information is stored in "web.xml". While exporting the SIP application, deployment descriptor is used to build the SAR file. Application server tool kit gives provisions to SIP deployment descriptor editor for the maintenance and creation of sip.xml descriptor file and also supports to web development descriptor editor to maintain and create web.xml.

For specify deployment information for modules, web deployment descriptor editor gives this provision and these modules are created in web development environment. And the information of modules will be appeared in web.xml file. Using web deployment descriptor can set web deployment descriptor attributes. The information, which is necessary for deploying web application module, is in web.xml file, which is also used for creating a WAR file. Web deployment descriptor includes many vies and also a dynamic, which defines many settings and properties while deploying descriptor.

SIP deployment descriptor editor has multiple tabbed pages like variables, references, security, servlet, source page and overview. We can use these pages as to get summary of contents or we can add, remove or change the contents. The overview page extends web deployment descriptor overview page. The irrelevant pages to SIP are removed and relevant pages to SIP are added, e.g. “login section”. The sections of overview page include general information, servlets, login, security, icons, listeners, references, environment variables and context parameters. Servlet page enable adding, removing or modifying of servlets in the project. Servlet page has the same sections as web deployment descriptor except some changes. And these changes are URL mapping section, which are replaced with mapping section for SIP. Also extension sections and programming model extensions are removed. Security roles and constraints of project are managed by security page. Security page of web deployment descriptor editor is extended by security page. In security page, web resource collection section is replaced by SIP resource collection section. The variable page enables to manage the list of listeners, environment variables and context parameters in the project. Reverences pages manage the project resource references. The references pages support remote EJB reference, local EJB reference, resource environment, and resource reference. The source page finally shows sip.xml source code.

d. Sample SIP Services

The capabilities of session initiation protocol are presented by many samples, and these samples are contained by application server toolkit. These samples are call-blocking sample, call forwarding sample, third party call control.

4.4.3 Hardware & Software Requirement for Application Server Toolkit

a. Hardware Requirement [9]

Intel Pentium III 880 MHz Processor (1.0 GHz Recommended)
512 MB RAM (Minimum), (1.0 GB is recommended)
900 MB disc space, and 100 MB TEMP disc space is needed during Installation
Display 1024X768 (Minimum)

b. Supported Operating system [9]

Windows XP
Windows 2000
Windows Server 2003
Red Hat Enterprise Linux 3.0
Red Hat Desktop Linux 3.0
SUSE Linux Enterprise Server (SLES) 9

4.5 IMS Enablement Kit

For getting access to core IMS enablers, IMS enablement tool kit adds more features in application server toolkit. These features describe the group list and presence from SIP servlet. The plug-in of IMS enablement promotes the J2EE development, java 2 platform to reinforce the foundation level creation properties of IMS applications. The enhanced J2EE focus on different perspectives, which are, **[9]**

- To accelerate the development of JSR 116 SIP servlets, it enhances the service Creation with new servlet wizards and SIP project.
- For the editing of SIP deployment descriptions like other J2EE components, it supports wizard and SIP archive (SAR) format.
- For the acceleration deployment to runtime servers, it packs the SIP components and J2EE.
- To decrease risk of errors, it compiles the automatic inclusion of SIP application server (SIP A/S).
- It has a gallery of SIP sample services.
- It has a specific help plug-in for IMS

The enablers like IMS enablers, location services; short message services (SMS) are used by the foundation level applications, which provides end-user encapsulated services. To make more rich IMS composite applications, foundation level applications gives the provisions of more interfaces.

4.5.1 IMS access gateways and enablers

The access gateways and enablers are, *Presence Server* which communicates to users and manages, distributes and collects real time information regarding user’s availability, access and willingness. In *Group List Server Component,* Network based groups are managed and created by administrators and users, and these users and administrators are provisioned by group list server. Permissions, access lists, other service-specific properties are maintained by group list server. In *IMS Connector* Industry leading application server platform is induced by some specific IMS-interfaces from IMS connector. This function is performed to deliver full IMS standards-complaint SIP application server. **[9]**

4.5.2 IMS Enablement Tool Kit Components

Following are the components of IMS enablement toolkit **[9]**

a. Diameter Resources

- Rf Interface Libraries: It provides diameter-messaging interface to IMS application server application. And the purpose is to run the application, so it may send offline accounting messages to billing or accounting servers.
- Sh Interface Libraries: The data of the user profile is updated and retrieved from home subscriber server (HSS). And the function is performed by Sh subscriber profile web services.

- Corresponding web services description languages (WSDLs) of Sh and Rf services: Through web services, interaction is made in-between IMS application server and user profile server. And due to this interaction, charging is performed. It also indices a function to notify the status of the specified user.
- Test clients of Rf and Sh.

b. Presence Resources

- Authorization APIs Library: Customized permission policies are developed by this liberary. These APIs also develop the plug-able applications. These plug-able applications allow or disallow, authorize and give information to subscription to presentity.

c. Parlay X.21 Resources

These are web services description languages (WSDLs) and include call notification, third party call, payment, SMS, terminal status.

4.5.3 Sample Application for IMS Foundation

There are three sample applications of IMS toolkit. One is IMS service control (ISC) SIP servlet and two diameter clients. These samples give demonstration of using diameter Sh and Rf clients and IMS service control (ISC) & session initiation protocol (SIP) servlet API. **[9]**

a. IMS Service Control (ISC) Interface Sample

It gives the presentation about the implementation of back-to-back user agent (B2BUA) service by using ISC and SIP servlet API.

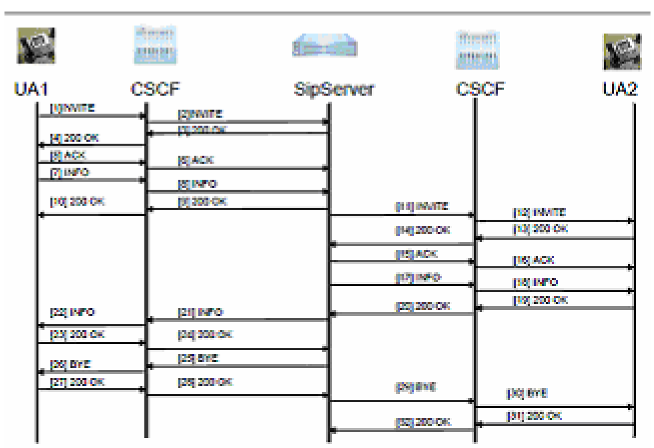

Fig: 4.2 ISC Demo Call flow [9]

The figure 4.2 shows the SIP server, CSCF (cal session control function) and interaction between user agents. User agent 1 sends a call to ISC demo SIP servlet and ISC demo SIP servlet receives INFO on same SIP session. This SIP session provides user agent 2 (UA2) address. Through serving call session control function (S-CSCF), ISC demo makes a call to user 2 and sends INFO to user 1 and user 2 separately. Then the call will be hanged in-between user 1 and user 2 by ISC demo. ISC demo is a SIP servlet, which is used to implement different methods like doInvite, doAck, doError, doBye, doRequest, doResponse, doSuccess, doInfo. IMS service control demo App (ISC demo App) holds the getters and setters and holds all the states. IMS service control demo App Handler (ISC Demo App Handler) used to handle the request. It performs some specific actions about the moving from current state to new state. The processing action INVITE from user 1 is also handled by IMS service control demo App Handler (ISC Demo App Handler).

b. Diameter Client Sample

There are two-diameter client samples.

- Diameter Rf test client, which uses offline charging WSDL DiameterRfService.wsdl
- Diameter Sh test client uses the user profile management WSDL DiameterShService.wsdl

In general both applications processing are similar in simulation and software configurations.

4.5.4 Hardware & Software Requirement for IMS Enablement Tool Kit

a. Hardware Requirements [9]

Intel Pentium III 880 MHz Processor (1.0 GHz Recommended)
512 MB RAM (Minimum), (1.0 GB is recommended)
900 MB disc space, and 100 MB TEMP disc space is needed during Installation
Display1024X768 (Minimum)

b. Supported Operating Systems [9]

Windows XP
Windows 2000
Windows Server 2003
Red Hat Enterprise Linux 3.0
Red Hat Desktop Linux 3.0
SUSE Linux Enterprise Server (SLES) 9

4.6 Web Services Server Toolkit

To create the higher level of services, web services server toolkit is used by the IMS composite applications. These IMS composite applications contain service enablers and foundation level applications. Service enablers and foundation level applications require principles of service-oriented architecture (SOA), or we can say that service enablers and foundation level applications acts as service oriented architecture (SOA) service implementations. New composite services are created when we plan these services and enablers by using BPEL. These new services are also service-oriented architecture (SOA) services and can create high level of services. For the composite IMS application, we need to install web service toolkit and we also require integration developer, which is powerful and integrated platform. **[9]**

4.6.1 Capabilities of Telecom Web Services Toolkit (TWSS)

One of the important components of telecom platform is telecom web services toolkit (TWSS). The capabilities and exposure of the network is enhanced and illuminated by technology and language independent high level web service interfaces. These interfaces may be access through custom integrated services, direct connect access to network protocols, SIP or Diameter, PSTN functionality through parlay. **[9]**

a. TWSS Service Implementation

These implementations have many reusable components, which are deployed at application server. They provide the best implementations and high-level service interfaces, which reinforces the network for convenient access of it.

b. WSS Access Gateway

It provides the capabilities like authorization, message capture, management, policy driven traffic monitoring. This is a gateway in-between service ends points and clients and this gateway implements such policies on all such requests and responses occurred in-between these service end points and clients. This gateway enhances the system capability by adding policy driven processing elements. It has mediation primitives, which are the components used to assemble customized message process flow. The programming model and mediation primitives interface creates the custom function and gives the points for extensibility.

c. Service Policy Manager

It provides the capability of access to service policies, data administration, access mechanism and definition of administration interfaces, storage. Attached policies, requesters and service definitions can be taken out by the usage of Service Policy Manager. Service Policy Manager also defines associated service subscriptions to requesters.

4.6.2 Mediation Service

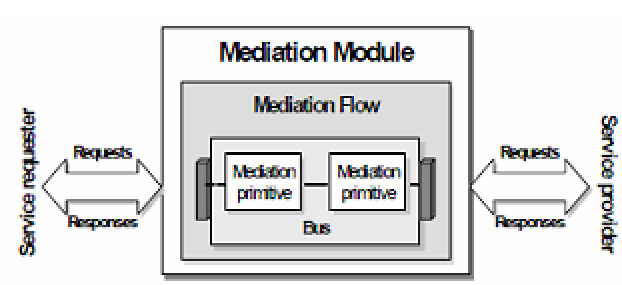

Fig: 4.3 Mediation Application Service [9]

The message processing between service providers and service requesters are allowed by enterprise service bus and mediation is a function of this enterprise service bus. By using mediation modules, these services are implemented and they modify and intercept messages passing in-between providers and requesters. The mediation modules in mediation flows, which process the messages, provide the logic. By using the mediation flow editor these flows are maintained as well as created. These flows are executed in a sequence and they consist of series of processing steps. The end nodes are translated by using these mediation flows and these nodes are based on source operation. For the creation of request and response flows, mediation primitives are added and help in execution of sequence between end-nodes. These request and response flows give the processing logic. The receiving of messages, processing them and send the processed message to next node is the main function of mediation primitives.

4.6.3 TWSS Access Gateway Mediation Primitives

The several mediation primitives are contained by telecom web services access gateway and they are divided into two steps. **[9]**

a. Mandatory Mediation Primitive

These are the primitives from the base of the TWSS gateway flow and configuration. The add-on mediation primitive function is supported and base services are provisioned. These primitives are:

- **Transaction Recorder Mediation Primitive**
  Within the table, the transaction information is recorded by the transaction recorder mediation primitive and is referenced by other mediation primitives
- **Policy/Subscription Mediation Primitive**
  Policy/Subscription Mediation Primitive retrieves the policy data that contains service, requester and operation being called. During the mediation primitive execution, policy data acts as decision parameter.
- **Service Invocation Mediation Primitive**
  Appropriate end points from the messages are extracted and further the message will be prepared for dynamic service invocation on the next step.

b. Optional Mediation Gateway

TWSS gateway flow uses following optional plug-ins:

- **Network Statistics Mediation Primitives:**
  The results of the exit information are stored in the database and the entries are entitled of records messages.
- **Service Authorization Mediation Primitive:**
  For web services operations it provides fine-grained authorization
- **SLA (Service Level Agreement) Enforcement Mediation Primitive:**
  It enforces the policies recommended by service level agreement and measure the system usage.
- **Group Resolution Mediation Primitive:**
  For the implementation of parlay-X services, we use resolution mediation primitive. for the given operation, these parlay-X implementations accept the group of URI's within the list of target destinations. The member URI's are replaced and expanded by the group URI's by Group Resolution Mediation Primitive.

- **CEI Event Emitter Mediation Primitive:**
  For the implementation of common base event (CBE), this primitive is used by alarm and fault common component. By this implementation, we can pick up fault information by different monitoring systems

4.6.4 TWSS Default Message Flow

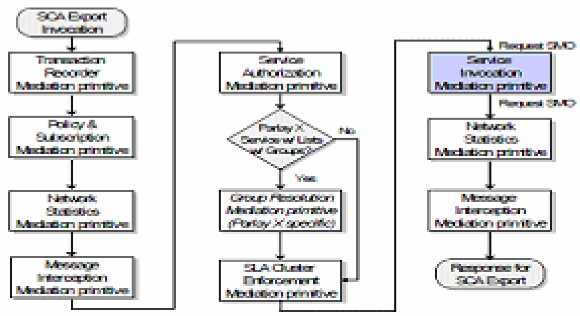

Fig: 4.4 TWSS Default Message Flow [9]

The default flow implementation is the provision of telecom web services access gateway. For different capabilities like traffic level enforcement, service/operation level authorization, message capture regulatory purpose and accounting of requests are supported by the default flow model, which is message processing function by a service provider.

4.6.5 Hardware & Software requirements for TWSS Toolkit

a. Hardware Requirements [9]

Intel Pentium III 1GHz Processor (Higher is Recommended)
1 GB RAM (Minimum 1 to 2 GB RAM is recommended)
5.5 GB of Disk Space
If file system is FAT32 and not NTFS, more space will be required.
1 GB for the TEMP directory is required
Display1024X768 Minimum (1280x1024 is Recommended)

b. Software requirements [9]

- Windows 2000
  - Windows 2000 Advanced Server With SP3 and SP4
  - Windows 2000 Server With SP3 and SP4
  - Windows 2000 Professional With SP3 and SP4
- Windows 2003
  - Windows Server 2003 Enterprise Edition
  - Windows Server 2003 Standard Edition
- Windows XP
  - Windows XP Professional With SP1 and SP2
- Linux
  - Red Hat Enterprise Linux 3.0 WS Update 2
  - SuSE Linux Enterprise Server 9

Chapter 5

IP multimedia Subsystems (IMS) Service Creation Tools

5.1 IMS service creation

For the service creation of IP multimedia subsystems (IMS), we need such kind of tools, which are helpful for an appropriate functionality of IMS. So far we have discussed different IMS toolkits, application server toolkit, IMS enablement toolkit and web service toolkit and their capabilities. In this chapter we will discuss different tools used by these toolkits for the service creation of IMS.

5.2 Web tools platform for AST (Application Server Toolkit)

For the building of J2EE and web applications, web tool platform provides such a platform for these applications. This web tools platform is an eclipse based project and shows the development of many tools and wizards. There are two standards of web tool platform, one is a Web standard tool (WST) and other is J2EE standard Tools (JST). The JST gives all the specifications of J2EE and these specifications are Enterprise Java Beans (EJB), Java Server Page (JSP), servlets and java web services. WST has all language neutral functionality for web applications building. WST also has document generators for languages like (SQL, Extendable Markup Language (XML), Hyper Text Markup Language (HTML)), validators, and editors. **[9]**

5.2.1 Categories of web tool platform

Within the development environment, *server tools* provide the management to J2EE servers. By the help of server tools, we can set up server connections, deploy and test applications and work with servers. These applications and activities can be viewed at the bottom of application server tool kit workbench. *Web Tools* provides creation of several web artifacts, which includes style sheets, Hyper Text Markup Language (HTML), java scripts and others; web tool gives the provision of many editors and wizards. Web tools also present TCP/IP monitor and embedded web browser. *XML Tools* provides validation and building of schemes, XML artifacts, Document Type Definition (DTDs), XML files, XML tools offer editors and wizards. *Web Services Tools* provides transformation of J2EE artifacts into other web services, web services tools gives the provision of web services wizards and Web Service Desription Language (WSDL) editor. *J2EE Tools* defines creation and working with J2EE artifacts, like EJBs and servlets, these tools represents a number of editors and wizards. J2EE also represents special editors for deployment descriptors. *Data Tools* gives an interaction with variety of databases. For the generation of Enterprise Java Bean (EJB) deployment code, which is added to many databases, are provisioned and supported by back end databases. **[9]**

5.3 Portlet Development Tool

These tools are compliant with the Java Specification Request (JSR) 168 portlet specifications. The creation of portlets and portlet projects are supported by these tools and they also presents portlet deployment descriptor editor. The generated portlets extends generic portlet class and they also have stubs for required portlet methods. By using the application server tool kit (AST), we can also import portlet Web Archive (WAR) files. Resources can be imported from a WAR file to our project **[9]**

5.4 Server Tool

By the help of this tool, we can publish any running instance of application server locally or remote. This tool improves support and defines the connection to data source and also gives the support for additional back end databases. The tool improves the Enterprise Archive (EAR) capability, which represents preparation and packing of applications for publishing application server. In the case of enterprise deployment descriptor, this improved EAR acts as deployment page. This tool is updated to give different provisions like adding of resource adaptors, message queues, connection factories and topics to an enterprise application on the target for the application server. This tool helps to launch profile management tool from the workbench and also helps to remove configuration and registry files, which are related with the workbench **[9]**

5.5 Telecom Web services Tools

Telecom Web services Tools are divided into two parts. One is an application-testing tool and the other is an application development tools. The application development tools consist of snippets, telecom application templates, cheat sheets, WSDL import wizard, and telecom web samples. The application-testing tools consist of simulator and runtime views and simulator configuration editor. **[9]** Parlay X2.1 WSDL Import Wizard is used to import parlay 2.1 WSDL into the existing project, we use Parlay X2.1 WSDL Import Wizard. Telecom Simulator Configuration Wizard is used for creation of new custom simulator configuration file, we use Telecom Simulator Configuration Wizard.

Using telecom application template sample, cheat sheets gives the provision and guidance for the creation of new applications. Each cheat sheet is addressed and designed to complete some tasks and it makes some steps with sequences to achieve the goal. These cheat sheets are designed to develop and test telecom web services. To develop and test parlay X 2.1 telecom client application, cheat sheet provides four steps. These four steps are:

- Create a web project: This step reinforces and helps throughout the process of loading the template of telecom web application
- Create the JSP servlet for your application: To call a telecom web service, this step guides the process thoroughly to create a client. This client may be a JSP or a servlet
- Add telecom web service call: by using the telecom snippets, this step guides the process to add web service call in the servlet client or JSP.

- Test your application: By using the web services client simulator, the step guides the process to test user client application. The web services client simulator runs on integrated application server in rational application developer.

Snippets are used to insert pre-defined working code in java server page or java class. These snippets are used to call different parlay X web services, which are multimedia media message service, short message service, Terminal location, accounting, payment, terminal status, third party call, audio call, notification administration, presence, group management, wireless access protocol push. The Categories of Snippets includes Telecom Message Service Category, Telecom Terminal Location and Status Web Service Category, Telecom Account Web Service Category, Telecom Notification Administration Web Service Category, Telecom Payment Web service Category, Telecom Call Web Service Category, Telecom Audio Call Web Service category, Telecom Group and Group Management Web Service Category, Telecom Member Web Service Category, Telecom Presence Web Service Category, Telecom Call Handling Web Service Category, Telecom Address List Management Web Service Category, Telecom Wireless Access Push Web Services Category.

The samples are used to make a toolkit that can be tested and deployed on web services client simulator. Web services client simulator provides the test suite and emulates the parlay X gateway to test the web applications of user development parlay X. And it does not needs a real network for this. For the configuration of test data, the simulator uses configuration file, which is xml.wss file and has static configuration file, which tells us the behave of the service.

5.6 Tools Developing & Testing SIP and IMS Sample Applications

5.6.1 SIPp

It is a free open source test tool/traffic generator for the SIP protocol. It is available under GNU general public license. This tool has a current version, which is 1.1rc5. It also represents some basic SipStone user agent scenarios like UAC and UAS. These tools read the SIP messages contained by XML scenario files and these files. These tools can describe different call flows leading from simple to complex. **[10]**

5.6.2 Ethereal

It is a network packet software analyzer, which is available for both Linux and Windows environments. This is a release under GNU general public license and open source software. It captures the network packets, which are flowing from or to the selected network interface and displays in real time with protocol independent information. It supports SIP protocol and represents filtering capabilities. **[11]**

5.6.3 SIPx Phone

It runs on Microsoft Windows and Linux. It is fully functional SIP soft phone. The phone client supports hold, multiple simultaneous calls, client mixed conferencing, mute, consultative transfer, authentication, multiple line appearances and extendable java based application environment. It is developed as open source and hosted as part of SIPX line projects. These projects are available from SIP foundry. It is licensed under LGPL. **[12] [13]**

5.7 Comparison of IMS traffic analyzing tools

So for we have discussed the IMS tools which are used for the service creation environment and the development of IMS applications. These tools are regarding to different tools kits like application server tool kit, IMS enablement tool kit, web services server tool kit and we have also discussed some other tools which help in IMS applications. According to our task we need to have some information in which e may analyze SIP message, time stamp and regarding information. For this purpose we study some tools which give information about the decoding of SIP messages and other useful functions. The name and capabilities of measurement tools for IMS are as under

5.7.1 DA 3400, DA 3600A VOIP Analysis [JDSU©]

a. Capabilities: [14]

- Real time analysis of 64000 simultaneous calls (DA-3600A)
- Real Time Analysis of 8000 simultaneous calls (DA-3400)
- Simultaneous VOIP and IP data Analysis
- MOS/R factor and detailed statistic for each call
- Extensive display customization and filtering
- Signaling analysis with call signaling trace
- Support post capture analysis and play back with PVA-1000

5.7.2 PVA-1000 VOIP Analysis [JDSU©]

a. Capabilities: [14]

- MOS and R factor Analysis
- Jitter and Packet loss analysis
- Audio playback with multiple CODEC support
- Signaling Analysis with call trace
- Signaling support for SIP, Cisco, SCCP, MGCP and DOCSIS/NCS
- Compatible with wide range of JDSU test and analysis equipment
- Compatible with Wireshark (PCAP)
- Distributed automated VOIP call capture agent option

5.7.3 Packet Scan [GL Communication©]

a. Capabilities: [15]

- Monitor progress of up to 500+ simultaneous calls with bidirectional RTP traffic
- Supported protocols-SIP,Megaco3525,Megaco3015,MGCP,H323/H225 and RTP
- Call capturing based on call agents or trigger actions such as MOS, R-factor, Jitter, packet loss, duplicate packets or called/calling numbers (SIP/H323)
- Provision for H263+ video capture and video conference monitoring capability
- Decode AMR in all packet format and G.726 RTP and AAL packing types
- Real time audio/video monitoring of RTP streams using audio playbacks record video and write to file features
- Call quality of service for all calls with E-model based G.107 MOS and R-factor with individual and summary statistics presented in graphical and tabular formats

- User can get real time call trace information based on SIP, H.323 calls
- Ability to configure sipport.ini file for customization of decoding options
- Calculate minimum, maximum and average RTD values for SIP calls
- Provide summary, detail, hex-dump and call detail records view of captured traffic
- Packet analysis displays call information in graphical format as well as in tabular format

5.7.4 Hammer Call Analyzer [Emprix©]

a. Capabilities: [16]

- Real time, multi stage, multi protocol call flow display
- Auto association of messages across signaling domains
- RTP media quality analysis, MOS scrolling for voice and video quality, save and play back
- Full VOIP protocol decode
- Intuitive, protocol aware searching, filtering and capture
- Pre-trigger packet capture
- Import standard libpcap traces for analysis
- Export standard format for documentation
- Analyze VOIP, IMS and TDM call using a single integrated analysis tool and interface

5.7.5 Hammer G5 [Emprix©]

a. Capabilities:[16]

- Multiple VOIP signaling protocols for IMS and NGN testing
- Emulate real end points with common, separate and unique characteristics for signaling and media
- Simulate real world subscriber behavior with pre-defined or customized caller profiles and load patterns
- Multiple codec and media types for wire line and wireless applications
- Active and passive verification and analysis of media quality
- Customizable protocol messaging and signaling behavior
- Remote access and control
- Comprehensive test monitoring and reporting
- Flexible test creation and execution options
- Secure transport testing

5.7.6 THGNOTEBOOK [Finisar©]

a. Capabilities: [17]

- THGnotebook features 7-layer packet decode and analysis, match including real time network statistics, advance alarm sating and action, multi state pattern, filters and automatic name table updating
- Provides a cost effective full power portable solution for measuring, analyzing or monitoring 10/100/1Gbit Ethernet

5.7.7 VOIP testing [Radcom©]

a. Capabilities: [18]

- VOIP testing analyzer used to capture, filter and analyze raw and decoded data on a wide variety of networks
- It decodes over 600 telecom and datacom protocols including standard protocols such as ITU-T, 3GPP and 3GPP2 and country/vendor specific variant as well including protocols such as SIP, H.323, RTP, Megaco and MGCP
- All layers of protocols are supported with VOIP testing

5.7.8 nGenius (Infinitestream) [NETSCOUT©]

a. Capabilities: [19]

- Packet troubleshooting recording
- Voice convergence management
- Data Granularity
- Network visibility
- Sophisticated Sniffer Intelligence analysis
- Response time Analysis/KPI
- Network Management Matrics
- Alarming and Event Identification

Chapter 6

Conclusion

For the past few years there has been a significant and growing demand for efficient as cost effective, secure and minimum response time multimedia services. The solution to this desperate need of customer is answered by wireless standard body 3rd Generation Partnership Project (3GPP) in form of/ as Internet Protocol Multimedia Subsystem (IMS). IMS converge voice, video, data and mobile network technology over IP base platform. IMS is designed to support the access the multimedia and voice services via wireless and wired station. IMS is not designed to standardize the application. IMS provide Quality of Service enabled multimedia services.IMS helps reduces time to market for rolling out new multimedia services it. IMS facilitate operator to charge the multimedia session appropriately

For fast, efficient and reliable IMS services, Quality of Service (QoS) is very important and challenging. Since IMS involves lot of signaling, the precision at signaling/protocol level is worthy. An efficient measuring/monitoring tool for traffic (signaling/data) will help researcher, designer, planner to enable QoS in order to obtain cost effective, secure and fast multimedia services.

The aim of the thesis is to identify a useful measuring tool for IMS signaling/protocol (Analysis). This tool attached with IMS test bed will extract information from the protocol elements which helps the researcher, designer and planner to observe the behavior of IMS signaling/protocol. Synthesis part of the thesis work will provide a proposal for appropriate existing measuring/monitoring tools based upon performance, cost, exportability and hardware requirement.

The comparitive details of IMS measurement tool is cited in chapter 5.7 based on the traffic handling capability, cost, exportability, performance and hardware requirement. PacketScan™ **[25]** is a real-time VoIP analyzer that captures live IP traffic, segregates them into SIP/H323 calls and collects statistics like timestamp etc by decoding the SIP/H323 packets. Hundreds of calls can be monitored in real-time including detailed analysis of selected voice band streams. Users can get an exact picture of QOS (quality of the service) and the technical adherence (adherence to the protocols specified by the standardizing authority) of the system under test.

PacketScan™ allows users to listen / record VoIP calls in real-time; perform power, frequency, spectral, tone and digit analysis with ease and precision. Its ability to monitor / record audio and video data of a session to files (in QuickTime *.qt format), allows to perform powerful video analysis. The captured VoIP calls with video can be played back using 3rd party VLC Viewer application.

The idea behind the traffic monitoring is provide high grade QoS to customer and Quality of Experience (QoE). For researcher, designer and planner the PacketScan™ by *Gl Communication Inc* is a optimum traffic analyzing tool that helps them to come up with efficient availability, reliability, performance of services to customer and efficient charging of multimedia session that helps the operator to charge customer for type of service they use.

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