COMSATS INSTITUTE OF INFORMATION TECHNOLOGY

Final Year Project Progress Report

Fixed-Mobile Convergence

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1 Introduction

Network convergence, or the combining of voice, data and video networks into a single system that uses Internet Protocol to process information, has become a hot trend within the IT industry during recent years. In the convergence network, however, VoIP signaling is handled in the packet switched networks so that the scalabilities can be easily improved through various computing technologies such as clusters. SIPcast is targeted for this movement. The Internet has experienced a dramatic growth in the last years. It has evolved from a small network linking a few research sites to a massive worldwide network. In the same time, cellular telephone networks provide services to over one billion users worldwide. The IP Multimedia Subsystem (IMS) is the technology that will merge the internet with the cellular world. It will make Internet technologies, such as the web, email, instant messaging, presence, videoconferencing, and multimedia applications available nearly everywhere. SIP is one of the most active initiatives underway in the IETF today and adopted by 3GPP as the signaling protocol of IMS. SIP is one of the most active protocols defined by IETF, which has many extensions. Development on the top of SIP needs good knowledge of SIP protocol.

1.1 IP Multimedia Subsystem

IMS is a subsystem that uses IP to communicate between different media. IMS is the architecture of the converged 3G/mobile Internet and SIP is the signaling protocol adopted by IMS. The IMS specifications define the functions to handle the signaling and user traffic for multimedia applications.The main signaling protocol in IMS is the Session Initiation Protocol (SIP). SIP is the proposed standard by 3GPP today for multimedia communication between users interacting with voice, video and instant messaging. In IMS, the use of SIP facilitates interconnectivity between fixed and mobile networks.

1.2 Session Initiation Protocol

SIP is the signaling protocol used by the IMS for Fixed-Mobile Convergence. SIP is the application layer control signaling protocol for creating, modifying and terminating sessions with one or more partcipants.It can be used to create two-party, multiparty, or multicast sessions that include Internet telephone calls, multimedia distribution, and multimedia conferences.It was originally designed by Henning Schulzrinne (Columbia University) and Mark Handley (UCL) starting in 1996.

1.3 Aim of Fixed-Mobile Convergence

Fixed-Mobile Convergence is aimed at providing a better and affordable convergence of fixed and mobile networks. It is based on the Handover between fixed and mobile networks like the Handover of GSM and Wi-Fi. FMC makes this handover possible and enriches the handover experience of the user. Now adays mostly cellular companies are providing GPRS and EDGE technology based on GSM system having low data rate but by this product the users can have a good data rate when they handover from GSM to Wi-Fi cell. So the product can attract the users and excell the market.

1.4 Goods about Fixed-Mobile Convergence

The evolution of internet from a small network of a few websites to an existing worldwide network and the increasing number of cellular users has motivated the idea of Fixed-Mobile Convergence.

1.5 Threats to Fixed-Mobile Convergence

Pakistan Telecommunication Authority may have concerns about the (Wi-Fi GSM) handover because of the tarrif belong to different operators. And a different network operator may not allow the (Wi-Fi GSM) handover.

The dual mode handsets are some expensive as compare to that of GSM compatible phones. Although it is not a very big issue so it can’t supress the great features of this project.

2 Methodology

In the designing process of the project we will get the help of IMS. As IMS best support the requirements and is actually designed for multimedia communication. For design the infrastructure of Fixed-Mobile Convergence we have to go through in details in the Layered Model of IMS.

2.1 IMS Layered Model

There are three layers of IMS. The Application, Control and the Transport Layer. The three Layers are as shown in the Figure below2.1.

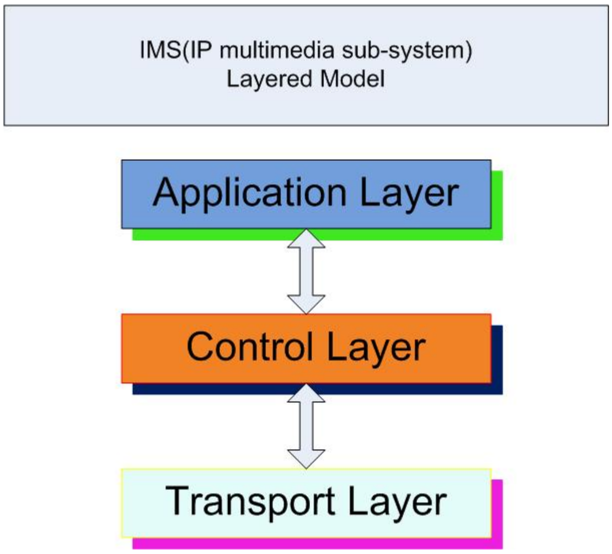

Figure 2.1: *IMS Layered Model*

The above three layers in the fig2.1 is the over view of the overall IMS structure. The IMS will be used as a support for convergence. The IMS itself doesn't give any new services but it

is a kind of architecture that provide the facility of including new services to the network.
The method which we adopt here is that a subscriber access IMS and its requests are defined and analysed according to the policy on the control layer and then its request is acknowledged from the networ which it wanted to access as per the routing process of the tansport and control layer.

3 Application Layer

The application layer hosts application and content services. This includes data centers of application server, web server, etc. Application Layer components and the signal flow at this Layer are shown in the figure below3.1

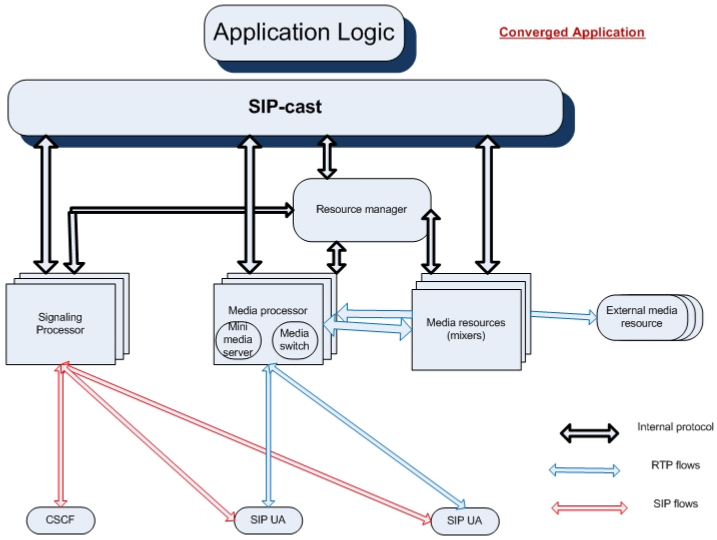

Figure 3.1: *API Module*

3.1 API Components

API is the platform on wich the application programs are build. The iportant components of API are described below.

3.1.1 Signaling Processor

Signaling processor is the most important part of SIPcast platform. It contains SIP stack and all SIP signals are processed here. Incoming SIP messages are translated into well-defined event to the SIPcast API module and a set of functions are provided for API module to customize the underlying stack and send out SIP messages. We can see the interfaces with other components including the SIPcast API module and resource manager in figure 3.1. When signaling processor starts up it registers its control port with the resource manager using a TCP connection. The status of signaling processor can also be reported to the resource manager using this connection. The API module requests signaling processor resource from resource manager. The resource manager select one signaling processor from all the signaling processors registered with it according to the load of each signaling processor. The control port of the assigned signaling processor is passed to the API module, and then the API module sets up a TCP connection to the signaling processor control port. All the upstream events and downstream commands are passed using this TCP connection. From figure 3.1, we can also see that only the signaling processor communicates with peers through SIP protocol. SIP signals may be sent to the signaling processor directly or be routed by CSCF(call state control function). One SIPcast application only needs one signaling processor to serve for it while one signaling processor can serve for multiple SIPcast applications. To make the SIPcast platform even more scalable, multiple signaling processor instances can be started and register with the resource manager. It enables load balance among the multiple signaling processors in the SIPcast platform.

3.1.2 Media Processor

The function of the media processor is the media processing. Media processor sends and receives RTP flows from SIP UA peers **??**. The media processing function can be divided into two parts. The first one is a minimedia server and the second one is a media switch. The embedded mini-media server can play an announcement, record user voice and detect users input while the media switch can route the RTP flows to an external media processing resources or platform integrated media resources such as mixers. The media switch acts as a RTP proxy so that in the view of the SIP peer, it talks with a consistent peer while the RTP traffic is processed by different servers. Similar with signaling processor, several media events are defined to indicate SIPcast API module the media processor event such detection of user input or the end of announcement. In addition, a set of media processing functions are provided for API module. From figure 3.1, we can see the interfaces with other components include SIPcast API module, resource manager or platform integrated media resources. Similar with the signaling processor, when the media processor starts up, it registers its control port with the resource manager using a TCP connection. The status of media processor can also be reported to the resource manager using this TCP connection. When the SIPcast API module requests media processor resource from the resource manager, the resource manager selects one media processor from all the media processors registered with it according to the load of each one. The control port of the assigned media processor is passed to the API module, and then the API module sets up a TCP connection to the media processor control port. All the upstream events and downstream media commands are passed using this TCP connection. One media processor can serve for multiple SIPcast applications. Multiple media processor instances can be started and register with the resource manager. It enables load balance among the multiple media processors in the SIPcast platform.

3.1.3 Media Resources

Media resources can either be the internal media resources or the external media resources as shown in fig3.1. These resources use the RTP flows routed by the media processor as the input media. After the specific media processing, out put media are transfered back to the media processor, which routed the returned RTP flows to the SIP peers. We can know there are three interfaces between media resources and other components. The first one is the registration channel with the resource manager. The second is the control connection with application and the last one is the RTP exchange with the media processor.

3.1.4 Resource Manager

The resource manager is used for the management of various resources in the SIPcast platform. The functions of the resource manager include resources registration, resources assignment and resources status monitoring. There are many kinds of resources such as signaling processors, media processors and mixers. More resources may be added to the current platform. When each resource starts up, it registers to the resource manager with its control port. It also uses the registration connection to report the status of the resources, for example the current users or sessions in the resource. This information can be used by the resource manager as a resource assignment criterion. The resource manager also provides an interface to query the status of the resources registered and set the policy for the resource assignment.

4 Control Layer

The control layer provides the session control and management, and responsible for setting up and taking down packet session. It also contains the Information of the subscriber, service authentication and location for which HSS has been installed on the control layer see fig4.1. This layer is also called the brain of the IMS because all the making of the generic policy decisions that are enforced on the Transport Layer are defined here at the control layer.
The control layer registers user device and route signaling messages between them. For this purpose SIP server has been installed on the control layer and this is called CSCF(Call Session Control Function) see fig4.1. CSCF interacts with the HSS(Home Subscriber Service), which is the database containing subsriber profiles and preferences.
The border gateways on the control layer provide the interoperabillit of different networks. MGCF(Media Gateway Control Function) and MRFC(Multimedia Resource Function Controller) communicate with CSCF and control the media gateways and media resources see fig4.1.

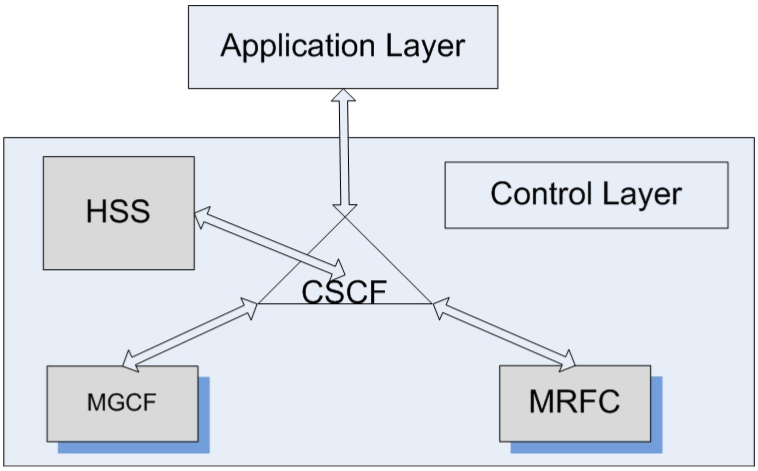

Figure 4.1: *IMS Control Layer*

5 Transport Layer

The need for the ubiquitous access is illustrated on the Transport Layer as in fig5.1. To take full advantage of the services, devices will access the network over many technologies such as cellulr, data, 3G, Wi-Fi, Wi-MAX, DSL, Fixed IP access, PSTN etc.

There is also an interface to connect the provider network to its internal Accounting and Billing System. The Media Gateways(MG) in fig5.1 provide the conversion between packet and circuit switched networks.

Routers and Switches exist on this layer. This is the layer where security firewalls,and optical transport reside, along with gateways translating between protocols and between packet and circuit based traffic.

The routing that occurs within the IMS transport layer is dependent on service quality variables such as Quality of Service (QoS), high availability (HA), and congestion and session management. The policies that set these features, are set in a higher layer (the Control Layer) and enforced in the Transport Layer.

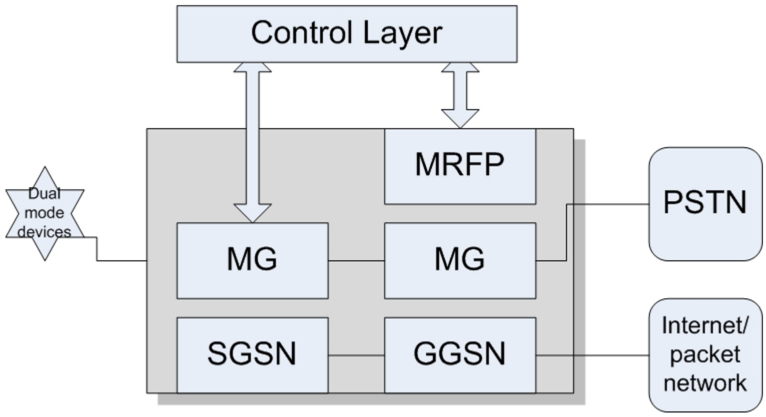

Figure 5.1: *Transport Layer*

6 Session Initiaton Protocol

The SIP(session initiation protocol) is used for the initiation of a session, termination of session or modifying a session during the communication take place. It can be used to create two-party, multiparty, or multicast sessions that include Internet telephone calls, multimedia distribution, and multimedia conferences. SIP is itself Text-based, allowing the humans can read SIP messages.
It is a light weight protocol that does the following.

- It provides mechanism for establishment of calls between a caller and a callee over an IP network. It allowsthe caller to notify the callee that it wants to start a call.It allows the participants to agree on the media encodings. It also allows the participants to end calls.

- It provide the mechanism for the caller to determine the current IP address of the callee. Users do not have a single, fixed IP address because they may be assigned addrersses dynamically (using DHCP) and they may have multiple IP devices, each with different IP address.

- It provides mechanism for call management,such as adding new media streams during the call,changing the encoding during the call,inviting new participants during the call,call transfer,and call holding.