A Project Report

On

Enhancement of the VoIPoEthernet interface for the Media Transport in Nokia Media Gateway 7520 IMS Voice Media Gateway

Submitted in partial fulfillment of the requirement for the degree of

Bachelor of Technology

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DECLARATION

I declare that this written consent impersonates my ideas in my own words and where others' ideas have been comprised, I have commended and referenced the native sources. I also proclaim that I have cohered to all propositions of academic integrity and have not forged or perverted or manipulated any facts/aspects in my consent. I understand that any infraction of the above will be root for the retaliation action by the institute and can also arouse punitive action from the sources which have not been properly commended or from whom appropriate approval has not been taken when required.

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Manisha Sharma(151004)

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Date - ____ May, 2019

CERTIFICATE

It is certified that the work contained in the project report titled "**Enhancement of the VoIPoEthernet for the Media Transport in Nokia Media Gateway 7520 IMS Voice Media Gateway**" by "**Sunayna and Rajat Bhushan**" has been carried out under my/our supervision and this work has not been submitted elsewhere for a degree.

Signature of Project Manager

Name

Signature of Supervisor

Name

JUIT, SOLAN

May of 2019

CERTIFICATE

It is certified that the work contained in the project report titled "Enhancement of the VoIPoEthernet for the Media Transport in Nokia Media Gateway 7520 IMS Voice Media Gateway" by "Manisha Sharma" has been carried out under my/our supervision and this work has not been submitted elsewhere for a degree.

Signature of Project Manager

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JUIT, SOLAN

May of 2019

ABSTRACT

Nokia Media Gateway 7520 give an End to End IMS (Internet Protocol Multimedia Subsystem) arrangement. It underpins open and progressively adaptable IP arrange framework that can oblige:

- Multiple traffic types: voice, information, video
- An assortment of access types: wired and remote
- Various Quality of Service (QoS) execution necessities
- An assortment of new and rising plans of action
- Mix of voice, information and video over broadband wired or remote frameworks

This voice, information media change must be executed while planning a start to finish engineering that will cost-viably control you toward an ideal cutting edge organizing condition.

Alcatel-Lucent offers redid arrangements based on IMS to address the novel needs of each specialist co-op as you manage the most recent correspondence patterns and the changing aggressive scene. While proceeding to offer improved premium VoIP administrations, the Alcatel-Lucent End-to-End IMS Solution centers around a client arranged methodology, so you can offer relational interchanges utilizing web, interactive media, informing and conversational administrations, client experience continuity crosswise over gadgets and systems, moment get to anyplace and self-administration personalization, and by conveying mixed way of life administrations to your shoppers and venture clients.

The Alcatel-Lucent End-to-End IMS arrangement is intended for fixed, portable and joined systems. It bolsters numerous entrance innovations, including Code Division Multiple Access (CDMA), Worldwide Interoperability for Microwave Access (WiMAX), Global System for Mobile Communications/Universal Mobile Telecommunications System (GSM/UMTS), and any computerized endorser line (xDSL).

ACKNOWLEDGEMENT

We owe our profound gratitude to our project supervisor Vijay Natarajan and project manager Sathis Kumar Venugopal who took keen interest and guided us all along in my project work titled “Enhancement of the VoIPoEthernet interface for the Media Transport in Nokia Media Gateway 7520 IMS Voice Media Gateway” helping us for our project by providing all the necessary information for developing the project. The project development helped us in learning and we got to know a lot of new things in our domain.We are really thankfulto him. His ethical help and direction gave us enough inspiration to work right. He has been kind and patient while proposing us the frameworks and basics of this venture and revising our questions and answering it all. We express gratitude toward his for his overall support.

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Date- ____ May, 2019

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CHAPTER 1

INTRODUCTION

Wipro Ltd is a main India based supplier of IT Services, including Business Process Outsourcing (BPO) administrations, internationally. The organization gives complete IT Solutions and Services, including Systems Integration, Information Systems Outsourcing, IT Enabled Services, Package Implementation, Software Application improvement and upkeep, and Research and Development Services to enterprises internationally.

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Product Name: Nokia Media Gateway 7520 IMS - UTRAN MGW

The Alcatel-Lucent End-to-End IMS arrangement is intended for fixed, portable and met systems. It underpins various access innovations, including Code Division Multiple Access (CDMA), Worldwide Interoperability for Microwave Access (WiMAX), Global System for Mobile Communications/Universal Mobile Telecommunications System (GSM/UMTS), and any computerized endorser line (xDSL).

While proceeding to offer improved premium VoIP administrations, the Alcatel-Lucent End-to-End IMS Solution centers around a client arranged methodology, so you can offer relational interchanges utilizing web, media, informing and conversational administrations, client experience con-tinuity crosswise over gadgets and systems, moment get to anyplace and self-administration personalization, and by conveying mixed way of life administrations to your shoppers and undertaking clients.

1.1 THEORETICAL BACKGROUND

Upgrade of the Voice over IP over Ethernet (VoIPoEthernet) interface for the Media Transport in Nokia Media Gateway 7520 IMS Voice Media Gateway.

The Voice over IP over ATM is utilized as an interface for Media Transport in the current framework. The IMS is associated utilizing UTRAN/BSS Radio Access Network on the Access Network.

Including Ethernet Interface helps associating the IPoEthernet or Wireless Network put together system with respect to the Access side system. It will empower a discretionary interface in the framework and furthermore help the designers on investigating any issues provided details regarding the ATM Networks.

1.2 PURPOSE OF THE PROJECT

Nokia Media Gateway 7520 give an End to End IMS (Internet Protocol Multimedia Subsystem) arrangement. Alcatel-Lucent offers tweaked arrangements based on IMS to address the extraordinary needs of each specialist organization as you manage the most recent correspondence patterns and the changing focused scene. The Alcatel-Lucent End-to-End IMS arrangement is intended for fixed, versatile and joined systems.

The Voice over IP over ATM is utilized as an interface for Media Transport in the current framework. The IMS is associated utilizing UTRAN/BSS Radio Access Network on the Access Network. Adding Ethernet Interface helps on interfacing the IPoEthernet or Wireless Network put together system with respect to the Access side system.

1.3 INFORMAL APPRAISAL

The work carried out in recent time have shown the benefits and increasing pace of using optical fibers and transmission of messages through it but it carries some disadvantages which inspired us to move towards the Ethernet usage and transmission of voice over Ethernet because of the use of technologies such as automation, computers , communication etc. which supports the cost effective and reliability like advantages.

With growing technology in recent years in the field of voice transmission, showing the potential and importance of communication and evaluating it with less error and more efficiency.

1.4 DESIGN STANDARDS

We have made this product very reliable and feasible with using the most easily available components. The design of the product is made keeping the convenience of user in mind thus resulting in a wireless compact model.

CHAPTER 2

ENTROPIA 3 ARCHITECTURE

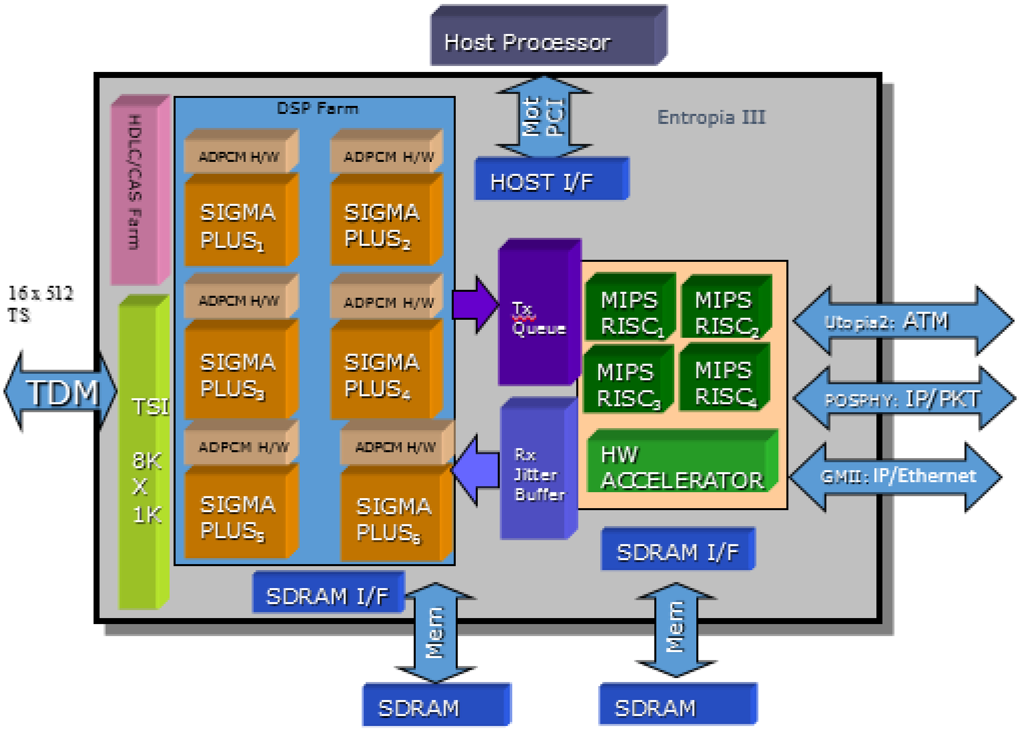

Fig 2.1 Architecture of Entropia 3

COURTESY : www.wipro.com

2.1 METHODOLOGY

Entropia 3 consists of two system on chips(SOC) where each one contains four RISCs and six DSPs. Through TDM interface with the help of GLCOMM, voice signal are transmitted. These signals are transmitted over DSP. For further processing, there are two buffers named as Jitter Buffer(JB) and Session Data Unit(SDU). Therefore, Voice signals are transmitted via DSP to RISC through SDU buffer for packetization and packets are transmitted via RISC to DSP through JB. JB and SDU are simply storage units which also helps to synchronize rate of transmission between DSP and RISC. Through RISC, packetization is done. Packets are directly transferred to Ethernet or the other way is to first sent the data over ATM protocol so that ATM cells can be formed. After that, ATM cells are transferred over UTOPIA interface to CISCO routers so that ATM cells can be converted into packets.

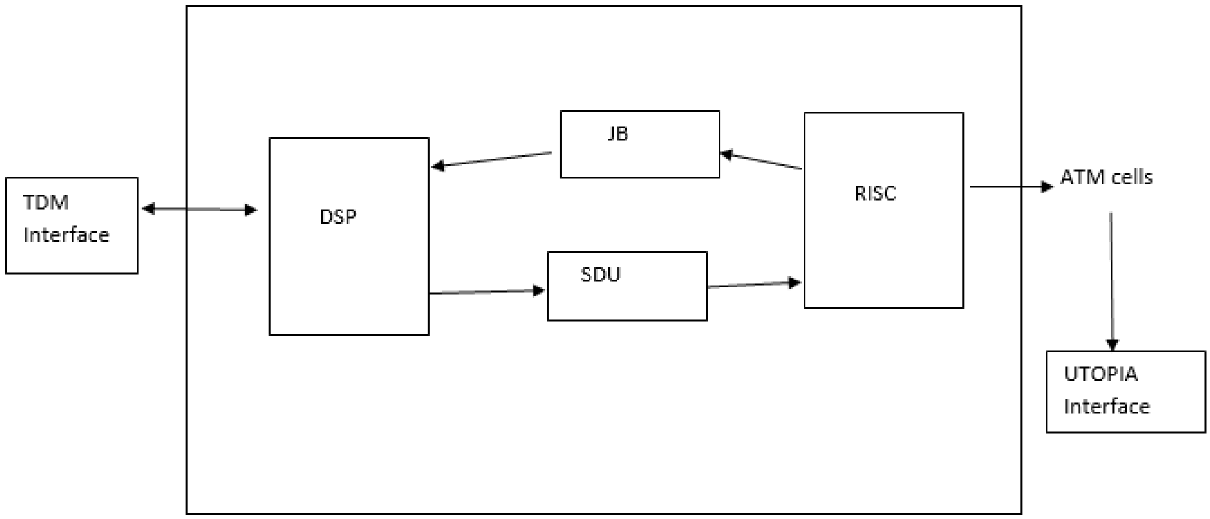

Fig 2.2 Working Model of Entropia 3

2.2 FUNCTIONALITY

There are three main functions of the Entropia 3 which is comprised of TDM world and packet world –

- Echo cancellation
- Compression
- Packetization

Echo cancellation and compression are performed through DSP with the help of sigma plus architecture and packetization is performed through RISC with the help of RISC architecture which works on the master slave concept and indulges AAL consequent services.

2.3 DSP

2.3.1 DSP ARCHITECTURE

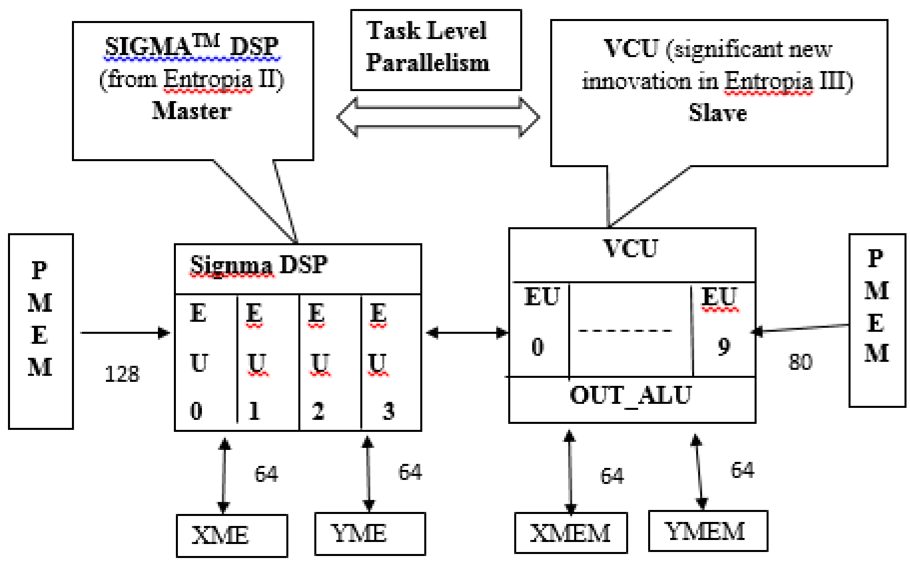

Fig 2.3 Architecture of DSP

- DSP architecture is vectorized VLIW core. Thus, it can process multiple scalar instructions concurrently. Although, data level parallelism is represented competently using vector instructions and processed on same parallel execution units of the architecture.
- Single instruction stream and have four execution data units.
- Each execution units contains 16 general purpose registers, two 40 bit ALSU units and 16*16 multiplier.
- Instruction packet size varies from 1 byte to 16 byte.
- Conditional execution unit and 8 address registers.
- Circular buffer with bit reversal support.
- Pipelining stages vary from mem-store(8) to alu.branch(11).

2.3.2 VCU ARCHITECTURE

- VCU architecture is again vectorized VLIW core.
- Single instruction stream and contains 10 executing data units.
- RISC like instruction set with conditional executing units.
- Each execution unit contains 16 general purpose registers, 16*16 multiplier, 32 bit ALSU and A-law/U-law PCM companding/expansion support.
- 11 input-output ALU to combine the result of 10 EUs.
- Instruction packet size is of 80 bit.

2.3.3 DSP-VCU INTERACTION

- VCU works as a slave of DSP.
- VCU has no interrupt capability.
- VCU memory is an extension of DSP memory.
- DSP-VCU interaction is done via VCU configuration registers.
- Parameter passing is via VCU task queue as well as common memory portions.

2.4 RISC

2.4.1 RISC METHODOLOGY

From DSP, voice signals are queued up in command queue CMDQ) and stored consequently in SDUQ buffer. Further, signals are stored in SDRAM of RISC0 (working as master) and then transferred to ATM segmentation and reassembly layer (ATM SAR). ATM cells are formed of fixed size of 48 bytes and cell segmentation is done via RISC1 (working as slave). Thus, these segmented cells are transferred to Entropia. Now, RISC3 (working as slave) performs packetization and packets are sent to RISC2 (working as master) so that it can be sent again over the DSP.

2.4.2 RISC ARCHITECTURE

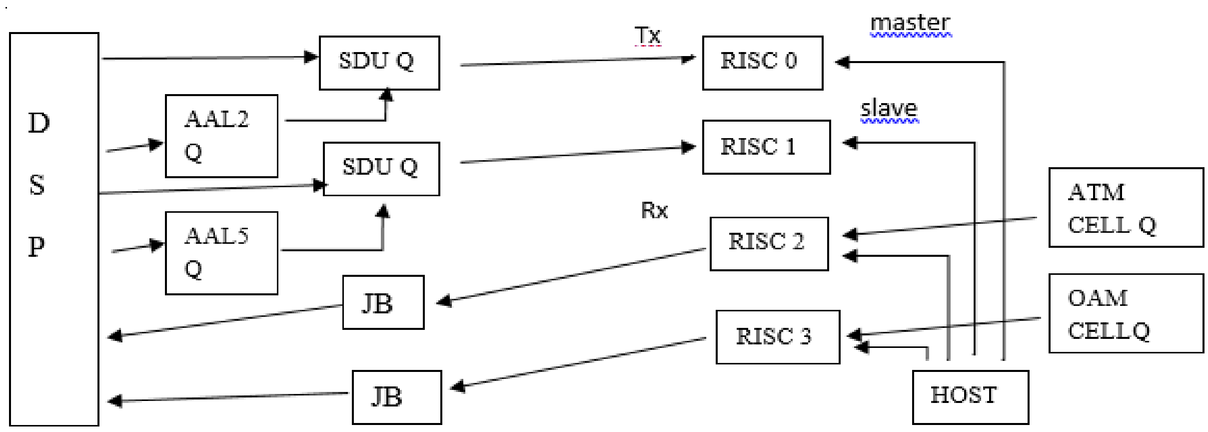

Fig 2.4 RISC architecture in Entropia 3

CHAPTER 3

ENTROPIA 3 GATEWAY

3.1 METHODOLOGY

There is media gateway which contains host, application and application peripheral interface. For TDM signals, HOST to DSP queue and DSP to HOST queue is maintained in SDRAM considered as DSP read and DSP write acknowledgement consecutively. Similarly, for packets, HOST to RISC queue and RISC to HOST queue is maintained in SDRAM considered as RISC read and RISC write acknowledgement consecutively.

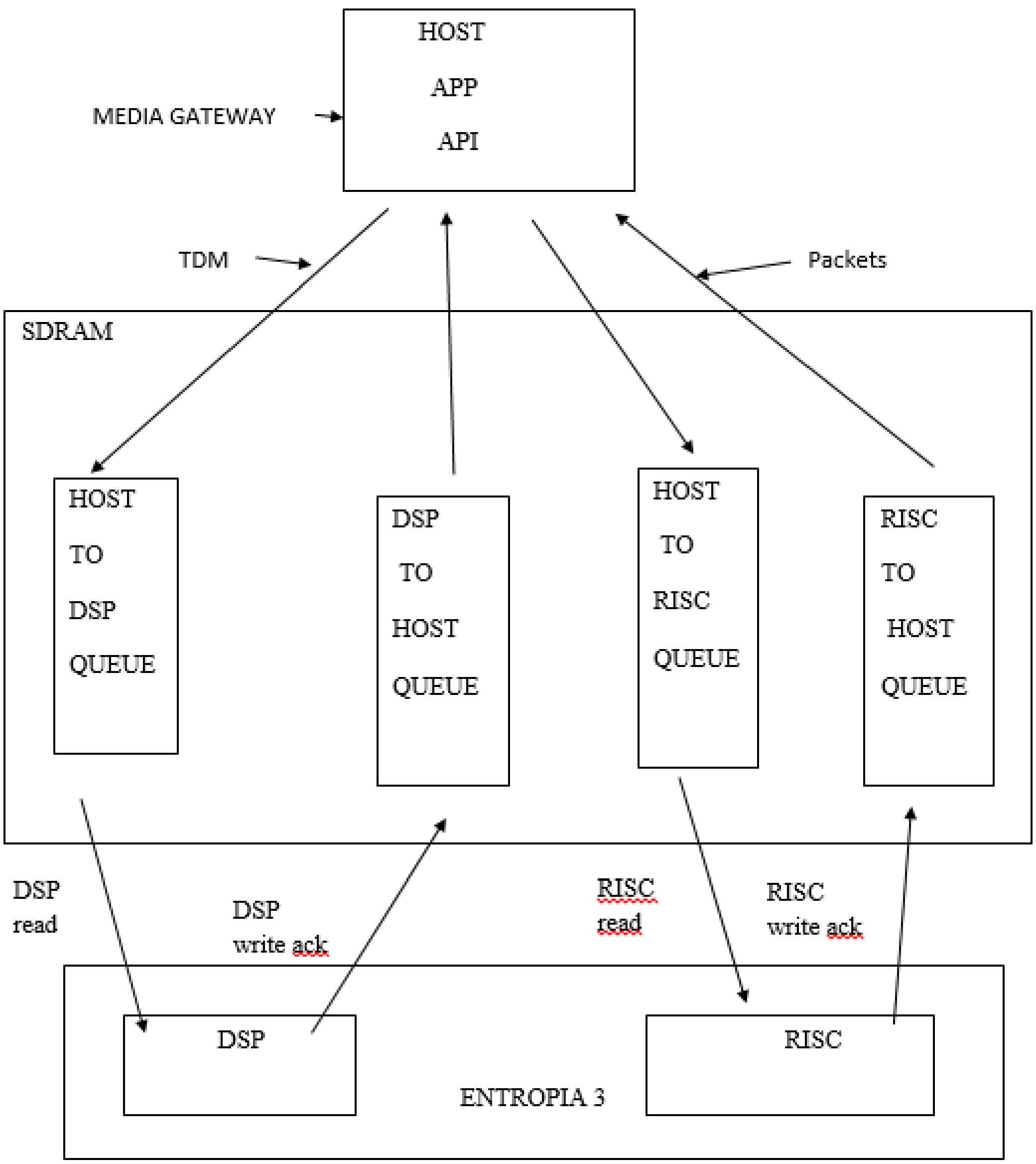

Fig 3.1 Entropia Gateway Flow

3.2 KEYTERMS

- GLCOMM is a hardware device acts as a source to transfer TDM voice signals to the Entropia 3 board via DSP architecture.
- WIRESHARK is a networking tool to analyze packet. Through WIRESHARK, all the information regarding the packets can be inspected, filtered, caught and monitored.

- CLI used for Entropia 3 is TERATERM. A script is made which exhibits all the details from booting sequence to the channel information of the board.

3.3 API CALL COMPLETION

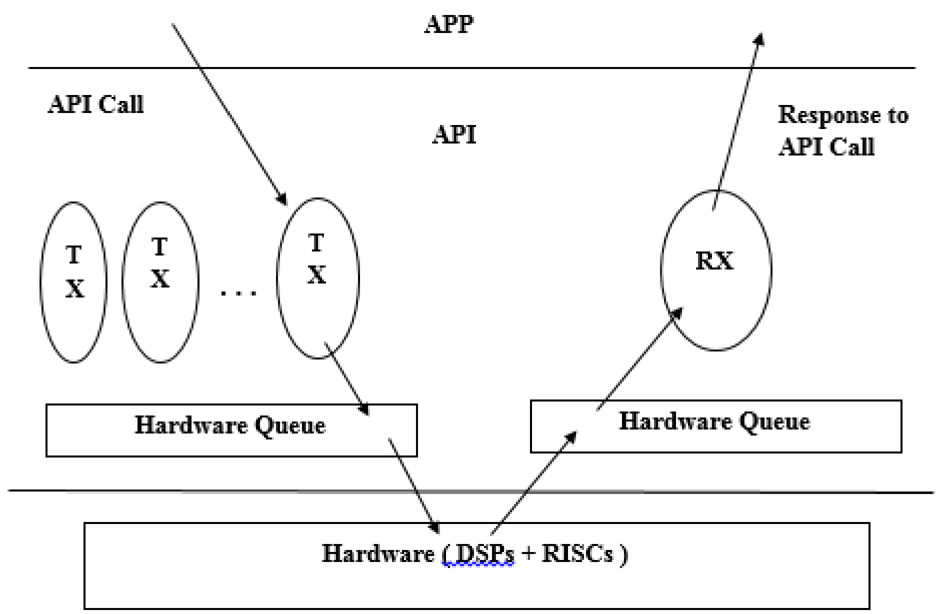

Fig 3.2 HLD Overview of API

Transmission (Tx) can be multiple instances of threads but receiving (Rx) will be single instance of thread.

The Entropia API library run-time model supports three method of calling the library functions –

1. Blocking Mode
2. Non-blocking Mode
3. Pseudo non-blocking Mode

3.3.1 BLOCKING MODE

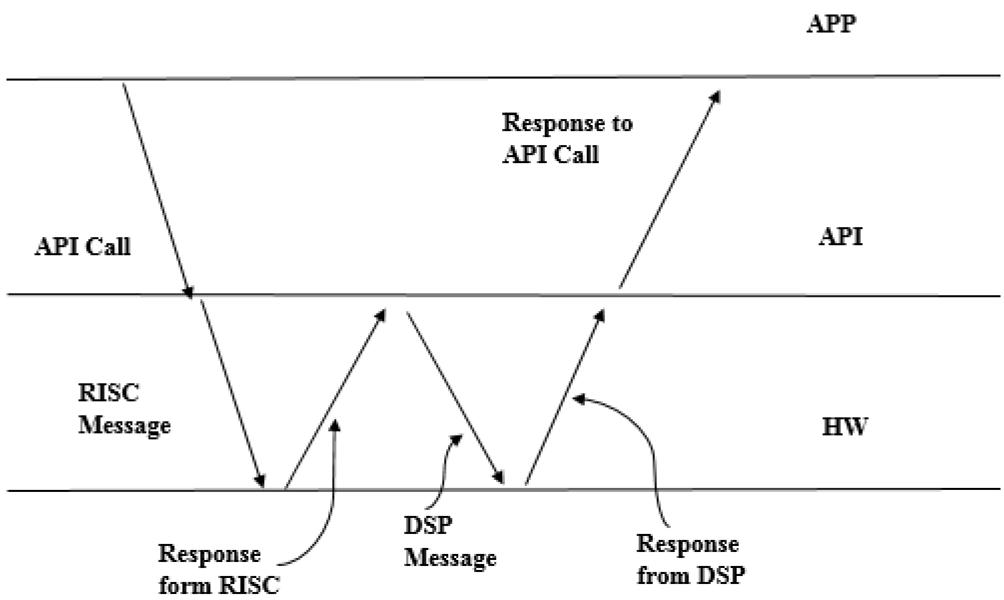

Fig 3.3 Blocking mode of API

In blocking mode, API sends response to APP for API call after receiving the response from both DSP and RISC.

3.3.2 NON-BLOCKING MODE

In non-blocking mode, API sends one response just after sending message to RISC/DSP immediately verifying all parameters are valid and another confirmation message to APP after receiving response from RISC and/or DSP.

Thus, blocking mode does not give any confirmation or acknowledgement about the message which API sends and it also depends on the response from both RISC and DSP.

Parameters validity is also processed by non-blocking method with the acknowledgement provided to RISC and DSP and depends on the either of the parameter.

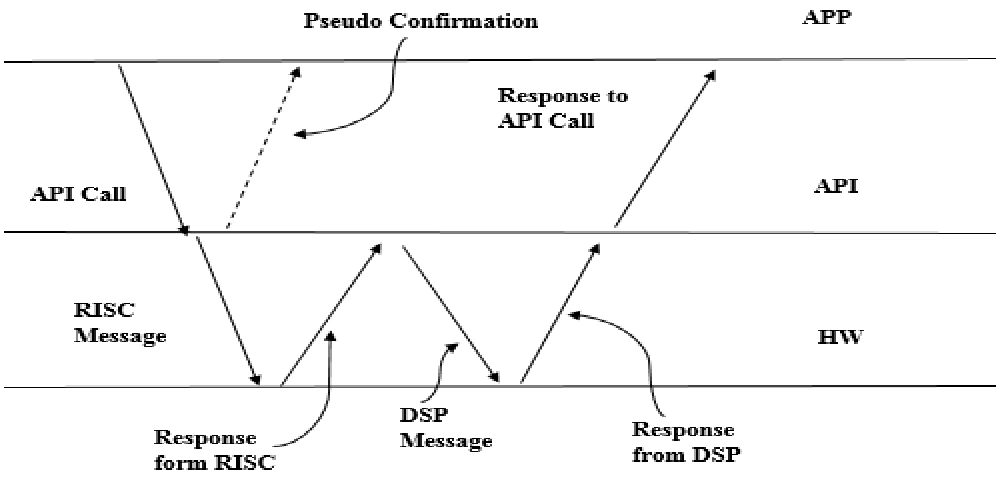

Fig 3.4 Non-blocking mode of API

3.3.3 PSEUDO NON-BLOCKING MODE

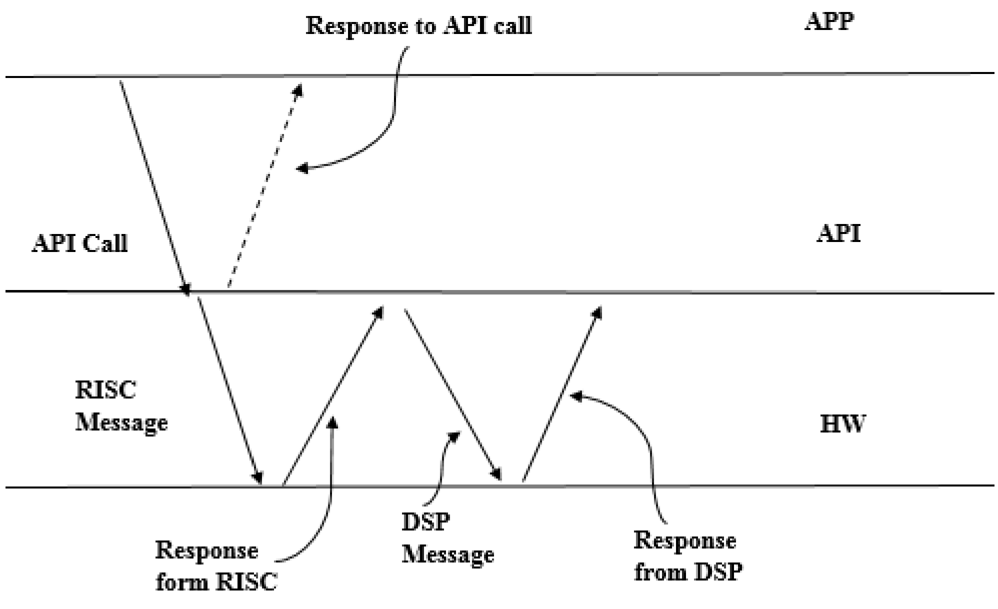

Fig 3.5 Pseudo non-blocking mode of API

- Meaningful if API sends response to both DSP and RISC otherwise it is same as non-blocking mode.
- Pseudo non-blocking mode is compile time option.
- The CLI setapimode 1 sets API to work in blocking mode with default time out time as 6 sec.
- If time out parameter to API call is zero, then API works in non-blocking mode otherwise it works in blocking mode.

3.4 API KEYWORDS

3.4.1 VOICE PROCESSOR

VP (Voice Processor) decides the chip on the assessment board. Chip is alluded as VP_. Entropia has two chips VP2 and VP3.

3.4.2 VP_HANDLE

- This is chip specific parameter
- The vp_handle is returned when the Entropia_DRV_Create_VP() is called and every further reference to the Entropia would be done dependent on this handle. Notwithstanding alluding to the Entropia chip, it additionally stores data explicit to the chip like:
  - DSP and RISC credits for each cores and the profile type information for each DSP
  - Hardware data explicit to the chip including the holding choice, the method of activity and the quantity of good DSPs
  - Mailbox IDs, start address of the mailbox pools of the API resources consumed
  - Channel insights for the VP
  - Support for declaration and recording highlights

- Stream ID and vacancy data for the dynamic channels
- Information on the parcel interface for which VP is arranged.
- Information on whether the various packet side features are enabled or disabled.

3.4.3 VCID

- VCID is part of VCP_HANDLE and each channel setup is related with an inside gained channel ID called as the VCID (Virtual channel ID)
- This VCID is procured by the API and educated both to DSP and RISC at the season of channel setup
- The SDU and Jitter Buffer location count by the DSP and RISC individually is dependent on the VCID data
- Used for showing DSP and RISC which SDU and JB supports to be utilized for specific channel
- The API pursues distinctive VCID distribution plans for various sort of channels (TDM-PKT channels, trans-coding channels and PKT-PKT channels).

3.4.4 Tokens

- Tokens are a collection of parameters grouped together for readability and software programming convenience
- Tokens are the input arguments for any API and by modifying the contents of the tokens the user can control the API feature
- TDM Tokens (voice, tone and announce)
- PKT Tokens (RTP, UDP and MAC/IP)

3.4.5 STREAM_ID

- It recognizes TDM_IN and TDM_OUT cradles for specific channel
- Maps schedule opening data to TDM (IN/OUT) cradles

3.4.6 TRANSACTION ID

- The Entropia API programming gives a nonexclusive system to following the advancement of the conjured API capacities, particularly for the non-blocking mode
- On every conjuring of an API work, the Entropia API programming allots a TRANS_ID (exchange ID) and returns this to the application code
- The application utilizes this TRANS_ID to inquiry the status of the pending solicitation API call utilizing the ENTROPIA_MSG_Confirm API call
- TRANS_IDs are novel in host area.

CHAPTER 4

OSI MODEL

Open System Interconnection (OSI) model consists of 7 layers having different functionality and usage for communication via different mediums. Communication is achieved via different protocols and their different activities performed in different layers with different approaches so that required communication in efficient time can be achieved.

4.1 PHYSICAL LAYER

- Processes transmission of bit streams between nodes.
- Media, signal and binary transmission.
- Coax, wireless, fiber, repeaters.

4.2 DATA LINK LAYER

- Processes transmission of data frames between nodes, connected by physical medium.
- Physical addressing is performed via MAC and LLC.
- Ethernet, bridge, PPP, switch.

4.3 NETWORK LAYER

- Manages routing and addressing among networking nodes.
- Path determination and logical addressing via IP.
- IP, ICMP, IGMP, IPSec.

4.4 TRANSPORT LAYER

- Performs segmentation and redirects acknowledgement. Data is transmitted between networking nodes.
- End-to-end connection and reliability is necessary.
- TCP, UDP

4.5 SESSION LAYER

- Interhost communication sessions are managed.
- Synchronization and data is sent to port.
- API, sockets, WinSock.

4.6 PRESENTATION LAYER

- Translation of data between networking services and an application.
- Data representation, compression and encryption is performed.
- SSL, IMAP, MPEG, FTP, SSH, JPEG

4.7 APPLICATION LAYER

- Terminal and directory services are managed (High-level API).
- End user layer and network process to application.
- HTTP, IRC, FTP, DNS, SSH.

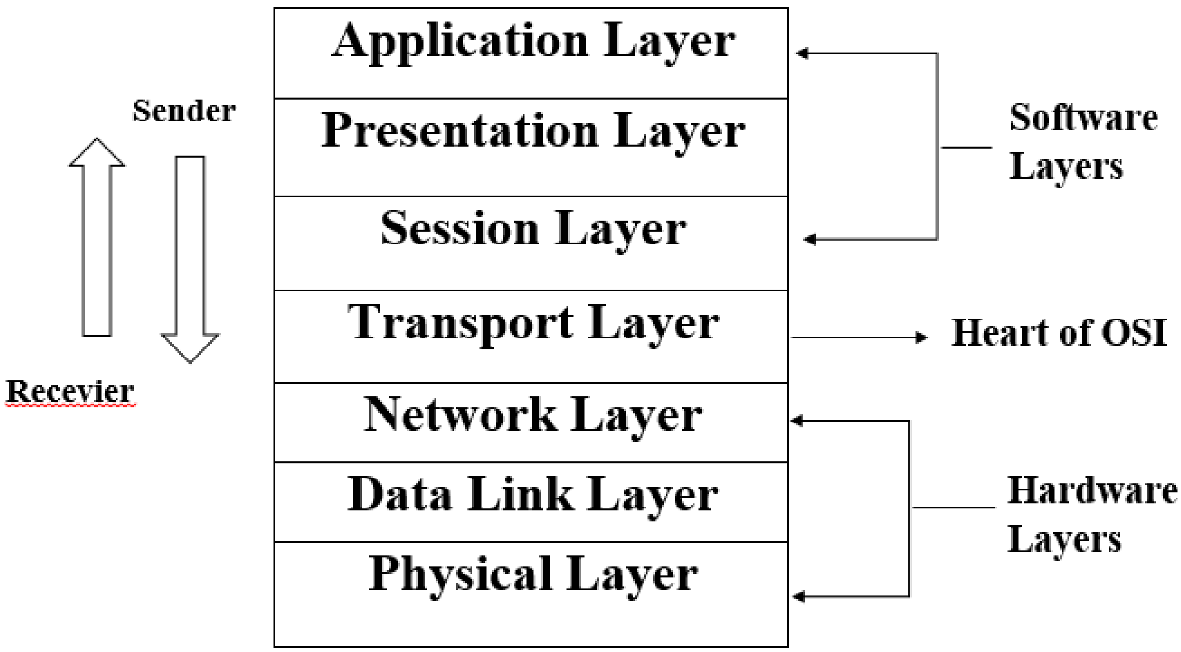

Fig 4.1 OSI model

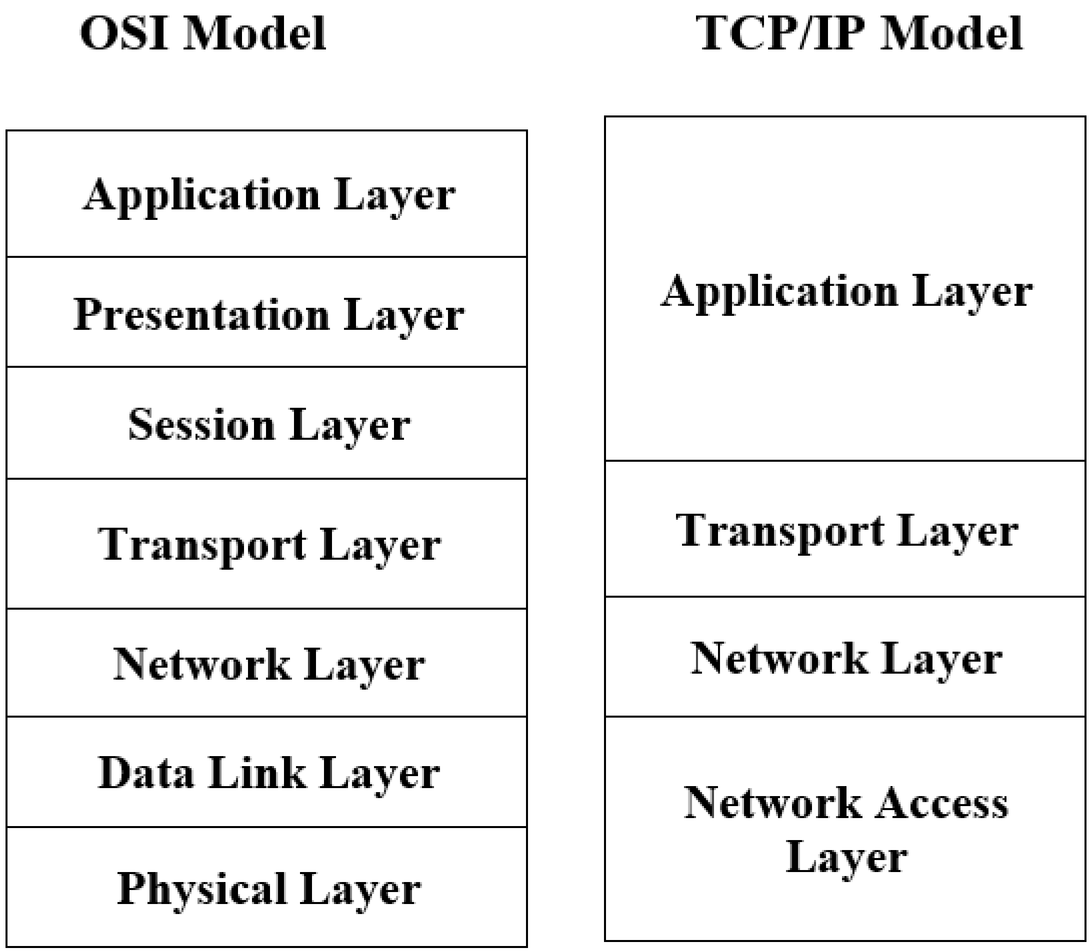

Fig 4.2 OSI vs TCP/IP Model

TCP/IP model is designed as for networking purposes as per the requirement of network requisites and procedures. It is older than OSI model and used as a standard model in lieu of reference model.

Physical and Data link layer is together known as network access layer used for communication through physical entities and addressing via MAC. Transport and Network layer functionality remain exactly same as of the OSI layer. All the translations are managed and all the servers and data services are maintained via Application layer.

CHAPTER 5

MEMORY OPTIMIZATION

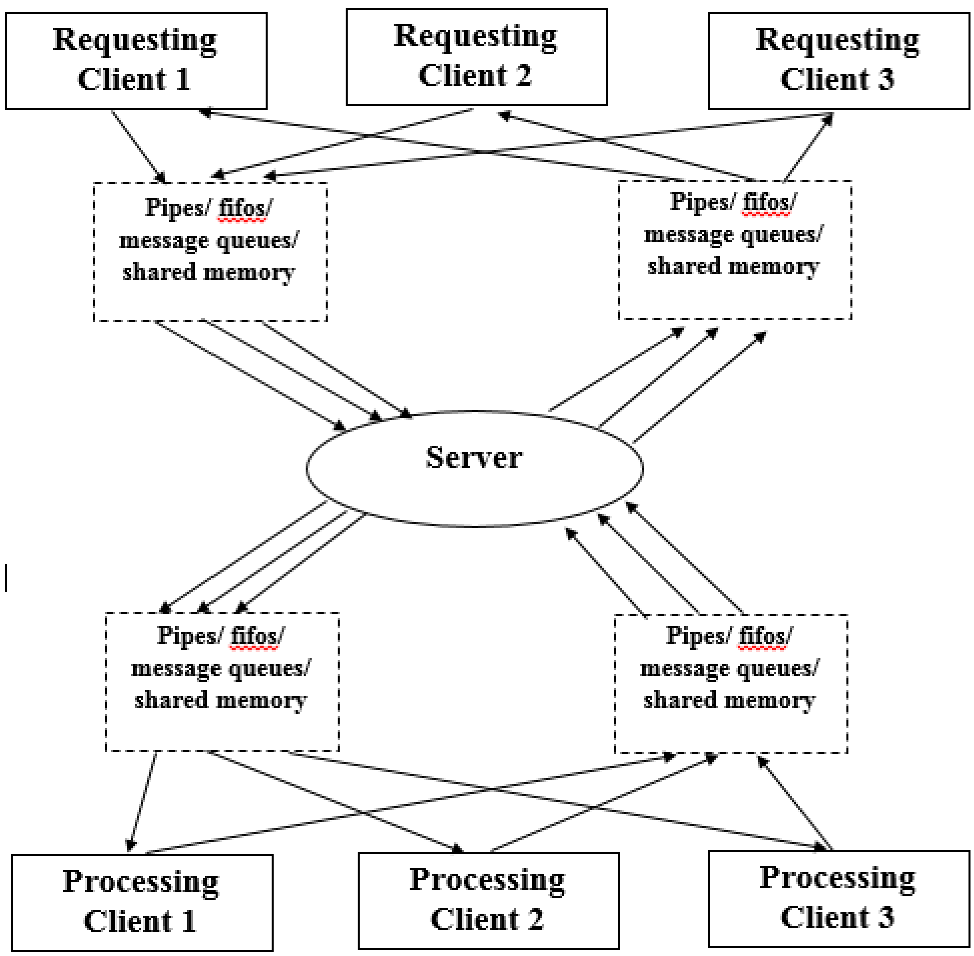

Fig 5.1 Communication Flow Diagram

There are 336 channels in Entropia board which are needed to synchronize and acts simultaneously. Initially, only 205 channels perform efficiently but with the help of memory optimization, channels that are synchronized and activated are from 292 to 310. There are various memory optimization techniques required for different purposes having different properties-

- Pipes
- FIFOs
- Message queues
- Shared Memory
- Semaphore
- Mutex
- Threads
- Spin Lock

5.1 PIPES

- Pipes are the simplest way for communication between two processes but processes must possess parent-child relationship.
- 64 KB buffer is maintained that is the size of buffer.
- Half duplex
- Sequential communication
- Pipes can not be accessed.
- Receiver end has no information about transmitter end.

5.2 FIFOS

- FIFOs can communicate between unrelated processes.
- 64KB buffer is again maintained that is the size of FIFO.
- Full duplex
- Sequential communication is not necessary.
- FIFOs are named pipes and can be accessed.

- Receiver end has no information about transmitter end.

5.3 MESSAGE QUEUES

- Message queues have message id and payload data so that receiver end can have message id of transmitter end.
- No buffer needed that is highly beneficial for memory.
- More synchronized and communication is far better.
- Faster communication
- Full duplex.

5.4 SHARED MEMORY

- Communication is even faster than receiving-transmitting speed as memory block is formed which is directly linked with process context block (PCB) of the other processes.
- Synchronization is more complex than that of previous processes.
- Full duplex
- Multiple read and multiple writes are possible.

5.5 SEMAPHORE

- Semaphore is atomic operation and takes many processing cycles during execution.
- Binary semaphore and counting semaphore.
- Semaphore is signaling mechanism.
- Semaphore is highly used to protect the shared resource from over utilization.
- 1 is for locking, 0 is for unlocking and negative means that the process or thread is locked.

5.6 MUTEX

- Mutex is mutual exclusion and takes lesser CPU cycles than that of semaphore.
- Mutex is locking mechanism.
- Mutex prevents race condition.
- Atmsot one thread can execute one section of code.
- Mutex has ownership unlike semaphore.

5.7 SPIN LOCK

- Spin lock is used only within one process at a time.
- Spin lock can not synchronize multiple processes.
- It tries to lock a particular process and until it is performed, it will wait or spin around the process.
- Spinning consumes high number of CPU cycles.
- Spin lock never switches context.

5.8 THREADS

- Thread runs within process so as it needs to use shared memory space.
- Multiple threads can acquired by a single process.
- Thread is smallest program instruction for execution.
- Threads can run simultaneously without complexity.
- Multithreading increases the assessing speed of the system.
- Thread has different libraries for execution in different space named as “pthread_library”.
- All the synchronization and signal entities are also different for threads.
- For same priority of threads, random selection is done by scheduler.
- Lesser CPU cycles are involved than that of processes.
- One thread can be started once till its end.

- Thread attains the most effective exertion of multiprocessing system.
- User level threads can be managed without the help of kernel.
- Kernel level threads are directly assisted and processed by operating system without any interference.

CHAPTER 6

PROTOCOLS INDULGED

6.1 TCP PROTOCOL

- TCP is reliable communication.
- TCP is Transmission Control Protocol.
- TCP is transport layer protocol.
- TCP works on the concept named three way handshake.
- TCP provides an acknowledgement.
- Packets are sent over TCP in sequential manner.
- There are six flags in TCP of size 1 byte each.
- Three main flags are SYN, ACK and FIN.
- Header size in TCP for IP packet is 20 bytes.
- State of communication is tracked by TCP segment via TCP header.

6.2 UDP PROTOCOL

- UDP is not reliable communication.
- UDP is user datagram protocol.
- UDP is transport layer protocol and faster than TCP.
- There is no guarantee for the delivery of packets.
- No acknowledgement is provided by UDP.
- Size of the UDP header is 8 bytes.
- UDP datagram contains UDP header and transported data.

- UDP checksum is 16 bit long.
- It is used for motionless and loss bearable connection establishment.
- Buffering is preferred in lieu of UDP for video streaming.
- UDP is mainly used for voice packets and live streaming.

6.3 RTP AND RTCP PROTOCOL

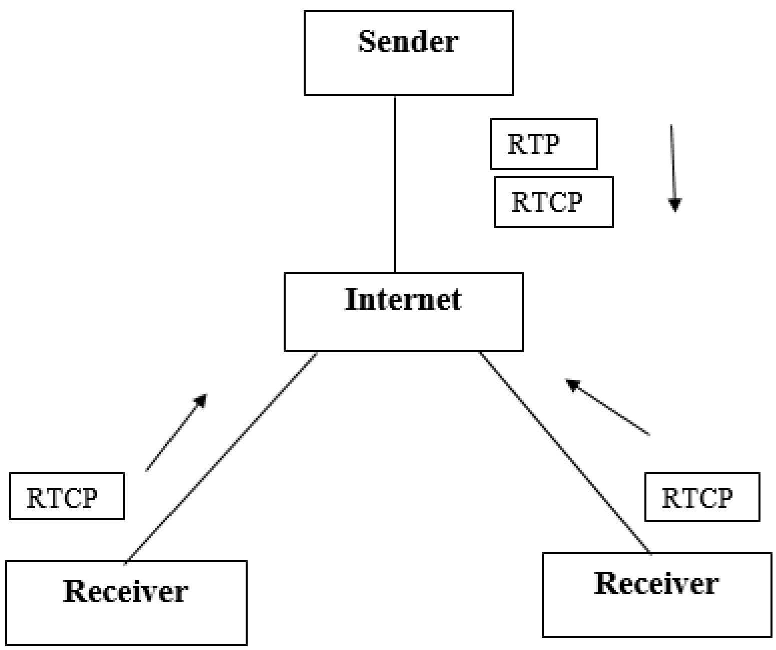

Fig 6.1 RTP and RTCP model

- RTP is real-time transport protocol and RTCP is real-time transport control protocol.
- RTP runs over UDP and works along with RTCP.
- RTP do not loose packets during transmission.
- RTP mainly performs exchange of audio-visual data on IP network.
- RTCP provides feedback on quality of transmission of multimedia data.
- RTP and RTCP applications are telephony, television and video conference.
- Media stream is only sent over RTP, transmission details provided by RTCP.
- Port range for RTP is from 16384 to 32767.

- Header size of RTP is 12 bytes.
- Framing over multicast is also processed by RTP.
- RTCP is also helpful in synchronization of multiple streams.
- RTP packets are designed at application layer and assisted to transport layer.

6.4 SIP PROTOCOL

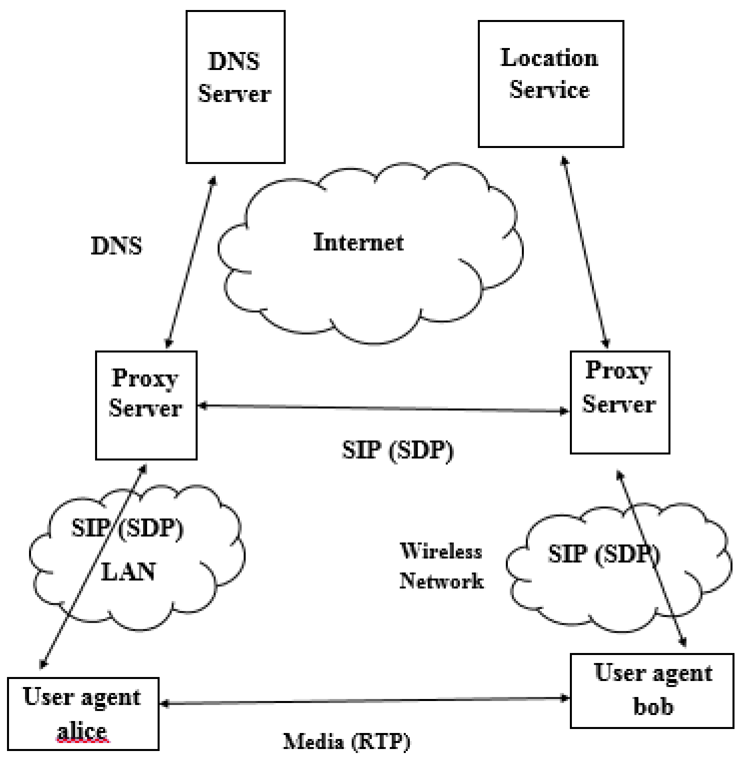

Fig 6.2 SIP Working Model

- SIP is Session Initiation Protocol.
- SIP is signaling protocol and always works in conjunction with other protocols.
- SIP is efficiently used for connection and disconnection of communicating sessions.
- SIP is an application layer protocol.
- SIP is preferred over PSTN and ISDN over internet.

- SIP trunks are used for simultaneous connectivity.
- Quality of transmission is not guaranteed.
- Packet-switched networking model is used for SIP trunk.
- SIP is preferred over other protocols due to having a virtual connection to PSTN.

6.5 ATM PROTOCOL

6.5.1 ATM MODEL

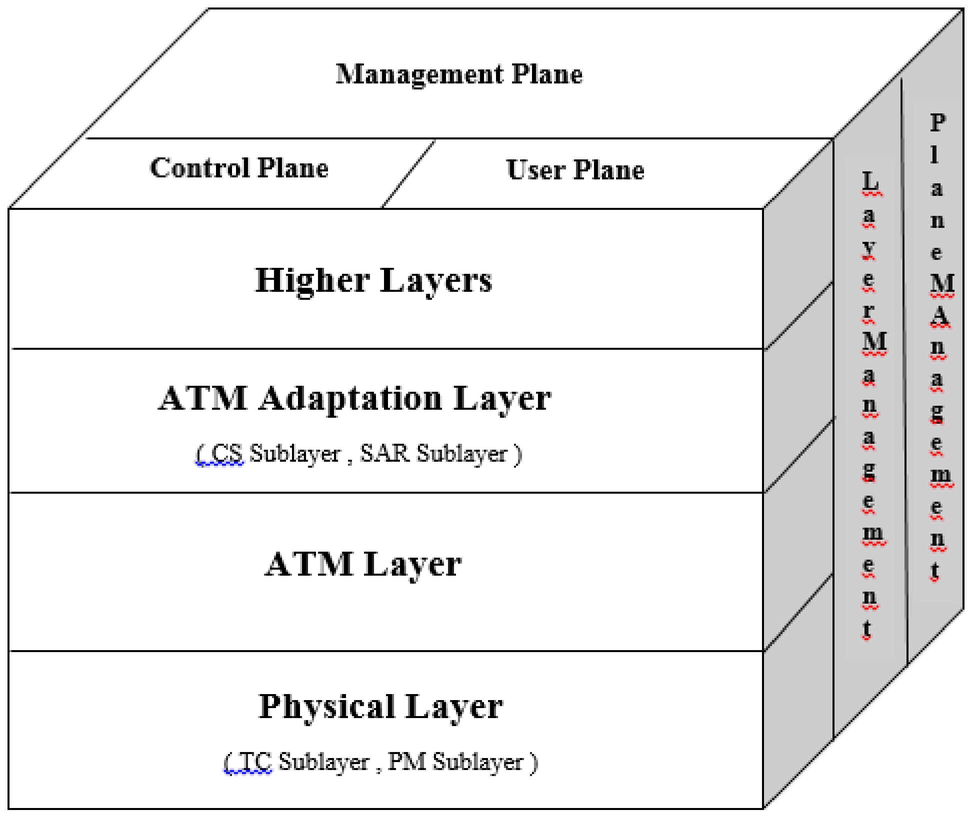

Fig 6.3 ATM Working Model

ATM protocol is Asynchronous Transfer Mode. With reference to OSI model, its physical layer is synchronized with physical layer of OSI model. ATM layer and ATM adaption layer (AAL) is directly synchronized with data link layer and similarly, upper layers are linked consecutively. ATM working model is also titled as ATM protocol stack. Each layer has different functionality having different attributes in correspondence with functionality management of the complete stack.

6.5.2 ATM FUNCTIONALITY

The functionality of ATM layer are classified into control plane, user plane and management plane where management plane is again classified into plane management and layer management. Each layer has its own activity to perform which is as follows –

1. User plane assists transfer of user information with associated controls. Control can either be error control or flow control.
2. Control plane assists control signals of calls and connections.
3. Plane management of management plane provides complete system functionality while the layer management of management plane provides synchronization of all resources and parameters of protocol entity.

6.5.3 ATM PHYSICAL LAYER

ATM physical layer is classified into transmission convergence (TC sublayer) and physical medium dependent sublayer (PMD sublayer).

Transmission convergence sublayer provides-

- HEC header generation and its verification.
- Cell depiction, presentation and description.
- Transmits frame recovery.

Physical medium dependent sublayer provides-

- Bit timing.
- Encoding-decoding of bits.

6.5.4 ATM LAYER

- ATM layer deploys the header of size 5 bytes to the segmented cell of 48 bytes each attained by AAL layer. ATM layer performs header deployment at transmit and receiver end.
- Generic flow is controlled by ATM layer.
- Virtual path identifier vs virtual channel identifier translation is processed.
- Multiplexing and de-multiplexing is also taken into account by ATM layer.

6.5.5 AAL LAYER

- AAL layer is classified into convergence layer (CS layer) and segmentation and reassembly layer (SAR layer).
- AAL performs encapsulation of user data designed by higher layers.
- Segmentation of data into size of 48 bytes at transmit host and reassembly of segmented data at receiver host is processed.
- AAL layer has multiple evolutions starting from AAL1, AAL2, AAL3/4 and at last AAL5.
- AAL1 and 2 both are connection oriented and used for circuit emulation application, video and voice conferencing.
- AAL2 supports also compressed audio and video format while AAL1 does not.
- AAL3/4 and AAL5 supports both connectionless and connection oriented data.
- AAL3/4 and AAL5 both are created for network service providers.
- AAL3/4 can not support directly IP packets while AAL5 can.
- AAL3/4 supports SMDS packets over ATM network.

- Cell generation in AAL3/4 is very complex while that is smooth with better cyclic redundancy check (CRC) in AAL5.
- AAL3/4 involves more steps than AAL5 for cell generation.

CHAPTER 7

SCRIPT

Following are the steps followed to bring up the chip after hard reset:

i. ‘@’(Auto boot) \ ‘c’ (to change the VxWorks image)
ii. TDM = 2
iii. svemain (1, 26, TDM, 1, 0, 0, 0, 1, 0)
iv. setapimode 1
v. eceb3

Each step is explained below:

1. Auto boot\ VxWorks image change
    - Enter ‘**@**’ for auto boot. For changing the VxWorks image enter ‘**c**’. Upon reaching the filename, specify the required image and continue.
    - VxWorks image will be downloaded through the TCP/IP interface using the warFTP application

2. To download the image into IDT,
    a. Open the WFTPD tool and go to **security** tab.
    b. Select **users/rights** option.
    c. Select the user name as **demo**.
    d. Select the path as **c:\download**.
    e. Press Done.
    f. Open the TeraTerm prompt and select the **setup** tab and go to **serial prompt** option.

g. Select the com port as COM1, baud rate as 9600 and transmit delay options as 1 and 350.
h. Select ok.

Now, restart the board by pressing the RESET button on our EVM.

Now we can observe the booting messages as below.

VxWorks System Boot

Copyright 1984-1998 Wind River Systems, Inc.

CPU: IDT S465

Version: 5.4.2

BSP version: 1.2/0

Creation date: Jun 8 2001, 08:52:24

Press any key to stop auto-boot...

6

[VxWorks Boot]:

[VxWorks Boot]: p (**p is a command for displaying the parameters**)

boot device : sn

unit number : 0

processor number : 0

host name : host

file name : Vxworks_3GN56452_Patch

inet on ethernet (e) : 10.11.12.2

host inet (h) : 10.11.12.1

user (u) : demo

ftp password (pw) : demo

flags (f) : 0x0

target name (tn) : eceb

[VxWorks Boot]:

[VxWorks Boot]: c (**c is a command for changing the parameters**)

boot device : sn

unit number : 0

processor number : 0

host name : host

file name : Vxworks_3GN56452_Patch **Vxworks_debug** (Here we are changing the file name.)

inet on ethernet (e) : 10.11.12.2 **^D** (Press cntl+D to stop changing.)

[VxWorks Boot]:

[VxWorks Boot]: @ (**@ is for booting the vxworks with Vxworks_debug**)

boot device : sn

unit number : 0

processor number : 0

host name : host

file name : **Vxworks_debug**

inet on ethernet (e) : 10.11.12.2

host inet (h) : 10.11.12.1

user (u) : demo

ftp password (pw) : demo

flags (f) : 0x0

target name (tn) : eceb

Here we can observe that the vxworks is booting with *Vxworks_debug* image.

Attached TCP/IP interface to sn0.

Attaching network interface lo0... done.

Loading... 9570603

Starting at 0x80010000...

Attached TCP/IP interface to sn unit 0

Attaching interface lo0...done

Adding 5561 symbols for standalone.

->

3. svemain
    - Engage initial system configuration

- Usage **: svemain**(bAutoMode, nSetNP, nSelectTDMmode, nSetPacketBusSize, nTdmLines, nSetPhysicalInterface, nSetEcebDiag, nSetTdmLineTrafficType, nSelectNPType)
- Options we uses are :

  1 -> nAutoMode

  26 -> nSetNP -> AAL2_AAL5 for all Voice Processors

  TDM -> nSelectTDMmode -> HMVIP_T1

  1 -> nSetPacketBusSize -> 16-bit

  0 -> nTdmLines -> 16 TDM lines

  0 -> nSetPhysicalInterface -> UTOPIA_INTERFACE - E3 ATM mode

  0 -> nSetEcebDiag -> diagnostics are bypassed.

  1 -> nSetTdmLineTrafficType -> VOICE_ONLY_MODE

  0 -> nSelectNPType -> select both AAL2 & AAL5 channels on 2 NP cores each

-> svemain(1,26,TDM,1,0,0,0,1,0)

--- svemain(): Engage initial system configuration ---

Options -- Prepare Adaptation Layer type per VP

1: AAL1 for all Voice Processors

2: AAL2 for all Voice Processors

5: AAL5 for all Voice Processors

9: AAL1t for all Voice Processors

13: ETHERNET for all Voice Processors

26: AAL2_AAL5 for all Voice Processors

Selected np for vp2: 26, and vp3: 26.

Options -- 0: GENERIC_T1, 1: GENERIC_E1, 2: HMVIP_T1, 3: HMVIP_E1, 4: GENERIC_8 MBPS

Prepare TDM_Carrier for: HMVIP.

Options -- Packet Interface Bus Size

0: 8-bit

1: 16-bit

2: LE_16-bit

3: POSPHY_LE_ODD_8BIT

4: POSPHY_LE_ODD_16BIT

Any other choice or press 'RETURN' for default to 16-bit.

Define Bus size: UTOPIA_16BIT_BUS_SIZE.

Options -- Number of TDM lines

0: 16 lines

1: 12 lines

2: 8 lines

3: 4 lines

4: 7 lines

Any other choice or pre ss 'RETURN' for default to 16 lines.

Prepare for 16 TDM lines

Options -- Physical Interface Type for Chip 2

0: UTOPIA_INTERFACE - E3 ATM mode

1: PACKET_INTERFACE - E3 POS_PHY mode

2: GMII_INTERFACE - E3 Gigabit Full Duplex mode

3: GMII_INTERFACE - E3 Gigabit Half Duplex mode

4: GMII_INTERFACE - E3 10/100bit Full Duplex mode

5: GMII_INTERFACE - E3 10/100bit Half Duplex mode

Any other choice or press 'RETURN' for default option 2.

Define Phy IF type: UTOPIA_INTERFACE

Options -- Physical Interface Type for Chip 3

0: UTOPIA_INTERFACE - E3 ATM mode

1: PACKET_INTERFACE - E3 POS_PHY mode

2: GMII_INTERFACE - E3 Gigabit Full Duplex mode

3: GMII_INTERFACE - E3 Gigabit Half Duplex mode

4: GMII_INTERFACE - E3 10/100bit Full Duplex mode

5: GMII_INTERFACE - E3 10/100bit Half Duplex mode

Any other choice or press 'RETURN' for default option 2.

Define Phy IF type: UTOPIA_INTERFACE

Options -- Prepare to Diagnostics

0, other choice, or press ENTER to bypass

1: Run Diagnostics APIs

diagnostics are bypassed.

Options -- Prepare TDM line for voice-only or voice-cas traffic

1: press 1 or ENTER for VOICE_ONLY_MODE

0: VOICE_WITH_CAS_MODE

Any other choice or press 'RETURN' for default to VOICE_ONLY_MODE.

Options -- select only AAL5 channels on 4 NP core OR both AAL2 and AAL5 channel s on 2 NP core each

1: Allows to select only AAL5 channels to be setup on all 4 NP cores

0: Allows to select both AAL2 and AAL5 channels on 2 NP cores each

select both AAL2 and AAL5 channels on 2 NP cores each

--- svemain: Initial system configuration completed ---

Type in eceb to initialize the system. Ready.

value = 1 = 0x1

->

4. setapimode
    - Usage **: setapimode(api_mode)**
    - api_mode = 1 for Blocking mode ; api_mode = 0 for Non-Blocking mode

In API we have 3 types of call completion modes, those are:

1. Blocking mode,
2. Non – Blocking mode and
3. Pseudo – Non Blocking mode.

Please go through the “**API_Call_Completion_modes.pptx**” for more details.

5. eceb3
    - Usage **: eceb3 (option)**
    - FPGA set up
    - Initialize VP API software
    - Memory Map Initialization
    - Run all BIST (DSP, TDM, RISC, PKT) for VP0 and VP1
    - Run SPRAM2 TEST
    - Download DSP code for VP0 and VP1
    - Initialize PKT_IF, QUEUE, SDRAM1, SDRAM2, TDM_IF for VP0 and VP1
    - Start 6 dsps for VP0 and VP1
    - All CO DSPs powered on.
    - Download RISC code for VP0 and VP1
    - Initialize HOST_IF, VP_RESOURCE, TDM_OUT
    - ----------Initialize VP API software----------
    - The VP installed on position: 2.
    - Entropia VP2 base address = 0xBF740000
    - The VP installed on position: 3.
    - Entropia VP3 base address = 0xBF780000
    - Multi Init VP Enabled.
    -
    - VP2_SUPPORT_AAL_TYPE = 26
    - Partial Init test enabled
    -
    - VP3_SUPPORT_AAL_TYPE = 26
    - Partial Init test enabled
    - >>RUN: ALL_BIST
    - DSP BIST Result for vp_idx (0): 0x3f
    - DSP BIST Result for vp_idx (1): 0x3f
    - TDM BIST Result[0] for VP_IDX(0):

- .... DONE !
- RISC BIST Result[0] for VP_IDX (0):
- .... DONE !
- PKT BIST Result[0] for VP_IDX (0):
- .... DONE !

CHAPTER 8

RESULTS AND CONCLUSION

8.1 RESULTANT WORKING

The IMS:

The Third generation networks aim to merge two most valuable resources in communication technology, along with local PSTN networks

- Cellular Networks
- The Internet

Description:

- Enhancement of the Voice over IP over Ethernet (VoIPoEthernet) interface for the Media Transport in Nokia Media Gateway 7520 IMS Voice Media Gateway.

- The Voice over IP over ATM is used as an interface for Media Transport in the existing system.

- The IMS is connected using UTRAN/BSS Radio Access Network on the Access Network.

- Adding Ethernet Interface helps on connecting the IP of Ethernet or Wireless Network based network on the Access side network.

- It will enable an optional interface in the system and also help the developers on debugging any issues reported on the ATM Networks.
- Several streams (voice and video both) can be sent in parallel with the same timestamp.
- The timestamp is used by the receiver to play the voice/video in a regular way for quality RTP is the media transport

8.2 NEEDS AND PROBLEMS

Optical fiber is ascending in both media transmission and information correspondence because of its unparalleled focal points: quicker speed with less lessening, less impenetrable to electromagnetic obstruction (EMI), littler size and more noteworthy data conveying limit. The endless transfer speed needs, then again, are additionally yielding noteworthy development in optical fiber requests.

Optical fiber systems work dependent on wavelength division multiplexing (WDM) innovation. WDM is a strategy, which utilizes more than one light source and identifier working at various wavelengths that at the same time transmit flag through a similar fiber while keeping up the message uprightness of each sign. You can do this insofar as there is a hole between every wavelength that is adequate to counteract any blending of the beats. This hole need not be extensive. Actually, holes in the scope of tenths of a nanometer are normal. Yet, we can see with the utilization of the optical fiber we run over issues like

- It is increasingly hard to introduce.
- Fiber optic link is little and conservative links and it is exceedingly defenseless to getting to be cut or harmed amid establishment and development exercises.

Furthermore, over the long haul utilization of optical fiber weaknesses of Optical Fiber which lead to search for some option:-

- Limited Application—Fiber optic link must be utilized on ground, and it can't leave the ground or work with versatile correspondence.

- Low Power—Light producing sources are constrained to low power. Albeit high power producers are accessible to improve control supply, it would include additional expense.
- Fragility—Optical fiber is somewhat delicate and increasingly powerless against harm contrasted with copper wires. You would be wise to not to contort or twist fiber optic links.
- Distance—The separation between the transmitter and beneficiary should keep short or repeaters are expected to help the sign.

8.3 CONCLUSION

We can conclude that the goal of the project is to enhance of the voice over IP over Ethernet (VoIPoEthernet) interface for the Media Transport in Nokia Media Gateway 7520 IMS Voice Media Gateway and the shift of 2g or 3g network to 4g network product according to the customer needs and requirements which may vary along the passing time and thus the product, Nokia Media Gateway 7520 IMS - UTRAN MGW will accommodate the new features of the technology and advancement.

8.4 FUTURE SCOPE

In future we can add more features to our project based on the latest technologies like we can embed all these different feature into our interface for media transport. Nokia Media Gateway 7520 give an End to End IMS (Internet Protocol Multimedia Subsystem) arrangement.

The controller capacities for both TDM and IMS/NGN systems. This item lets specialist organizations relocate flawlessly from inheritance TDM, to VoIP, to joined IMS on a solitary stage.

Specialist organizations can coordinate the flagging passage usefulness inside a similar body. The MGC-8 bolsters four key applications:

- NGN Class 4 neighborhood, access, and toll-couples
- CDMA/GSM Gateway MSC (GMSC)

- IMS Media Gateway Control Function (MGCF)
- Interconnection Border Control (IBCF)

It bolsters open and progressively adaptable IP organize foundation that can oblige:

- Multiple traffic types: voice, information, video
- An assortment of access types: wired and remote
- Various Quality of Service (QoS) execution prerequisites
- An assortment of new and developing plans of action

SUMMARY

In this project, we choose to work on Enhancement of the voice over IP over Ethernet (VoIPoEthernet) interface for the Media Transport in Nokia Media Gateway 7520 IMS Voice Media Gateway helping the transfer for voice transmission. This project is completed under the guidance of Satish Kumar Venugopal. This will help them for betterment of the cost effective and increasing reliability. And for this to happen we have shifted from 2g or 3g to 4g which will support the advancement of the technology. As with rapid change in the technology the product Nokia Media Gateway 7520 IMS - UTRAN MGW will accommodate the new features of the technology and advancement as per customer needs and updated requirements time to time which can vary along but transformation of channelizing the information of voice packets over the Ethernet which is now the main concern and to be worked on and still work is in progress. It is supporting the different access advancements, including Code Division Multiple Access (CDMA), Worldwide Interoperability for Microwave Access (WiMAX), Global System for Mobile Communications/Universal Mobile Telecommunications System (GSM/UMTS), and any computerized supporter line (xDSL) and much more according to the customer needs and requirements time to time.

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