A common exchange approach provisioning connection of existing SS7 with IP based SIGTRAN network using CPLD driven device

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Abstract-Now-a-days communication engineering is facing an age of explosive transformation that is both facilitating and influencing the unification of services. Compared to voice, packet data is becoming more significant proportion of traffic. It has become essential to find ways of consolidating voice, data traffic, platforms and services to reduce the operational, maintenance and initial cost of the network. IP based SIGTRAN network is now considered the most promising media on which to build the new integrated services platform which is basically SS7 in the IP packet. As a part of on-going integration of circuit networks and SIGTRAN networks, we have proposed a very cost effective common exchange developed by using CPLD driven device to consolidate existing SS7 network and SIGTRAN network. Our exchange comprises of hardware driven by CPLD, E1 card and ASTERIX software. For SS7 and SIGTRAN protocol configuration, ITU-T and IETF reference documents defined structure is used, which makes the data exchanger software format globally accepted.

Keywords: *SS7, SIGTRAN, CPLD, Asterix, E1*.

1. Introduction

SS7 or Common Channel Signaling System No. 7 is a global standard for telecommunications defined by the International Telecommunication Union (ITU) Telecommunication Standardization Sector (ITU-T) [1]. The standard defines the procedures and protocol by which network elements in the public switched telephone network (PSTN) exchange information over a digital signaling network to effect wireless and wire line call setup, routing and control. SS7 messages are exchanged between network elements over 56 or 64 kilobit per second (kbps) bidirectional channels called signaling links. Signaling occurs out-of-band on dedicated channels rather than in-band on voice channels [2].

The SS7 protocol stack borrows partially from the OSI Model of a packetized digital protocol stack. OSI layers 1 to 3 are provided by the Message Transfer Part (MTP) of the SS7 protocol; for circuit related signaling, such as the Telephone User Part (TUP) or the ISDN User Part (ISUP), the User Part provides layers 4 to 7, whereas for non-circuit related signaling the Signaling Connection and Control Part (SCCP) provides layer 4 capabilities to the SCCP user [3].

The MTP covers the transport protocols including network interface, information transfer, message handling and routing to the higher levels. SCCP is a sub-part of other L4 protocols, together with MTP 3 it can be called the Network Service Part (NSP), it provides end-to-end addressing and routing. TUP is a link-by-link signaling system used to connect calls. ISUP provides a circuit-based protocol to establish, maintain and end the connections for calls. TCAP is used to create database queries and invoke advanced network functionality mobile services (MAP), etc. [4]. The SS7 protocol stack is shown bellow:

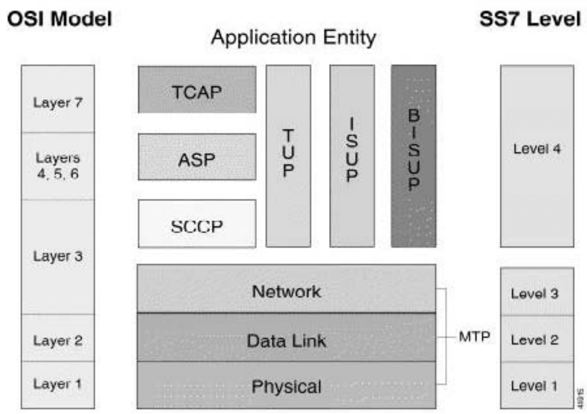

Fig 1. SS7 Protocol Stack [5]

Each SS7 data is transmitted in MTP (Message Transfer Part). Each MTP contains different SS7 layers, which are well defined by ITU-T. The MTP structure for SS7 data is like,

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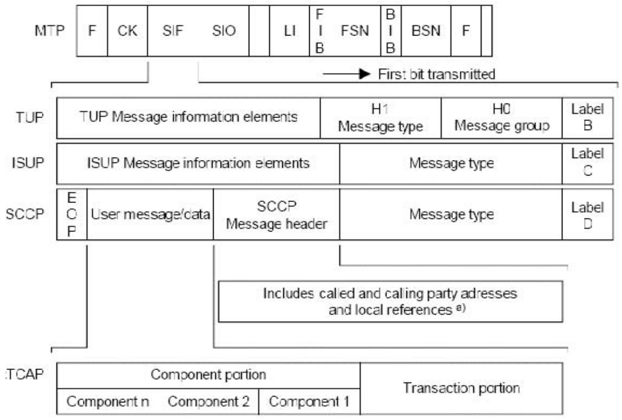

Fig 2. MTP structure of SS7 data [6]

The network architecture of SS7 network is shown bellow:

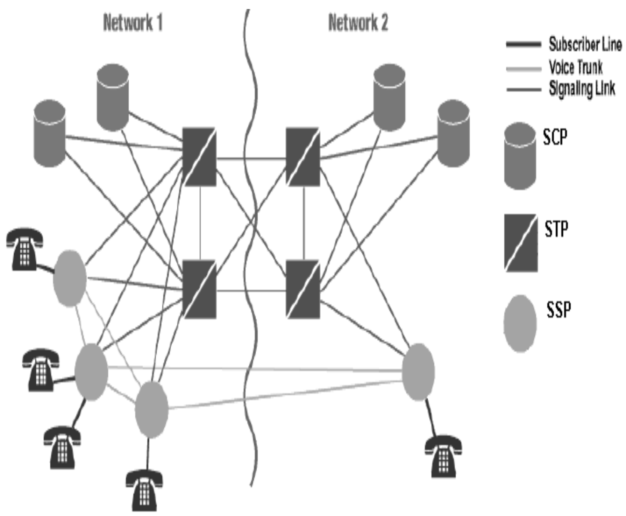

Fig 3. Existing SS7 network interconnection

As data has become significant proportion of traffic like voice, SIGTRAN is used for consolidating voice and data traffic. Signaling Transport (SIGTRAN) is a new set of standards defined by the International Engineering Task Force (IETF) [7]. This set of protocols has been defined in order to provide the architectural model of signaling transport over IP networks.

The SIGTRAN protocols specify the means by which SS7 messages can be reliably transported over IP networks. The architecture identifies two components:

1. A common transport protocol for the SS7 protocol layer being carried and
2. An adaptation module to emulate lower layers of the protocol.

The SIGTRAN protocol stack is show bellow:

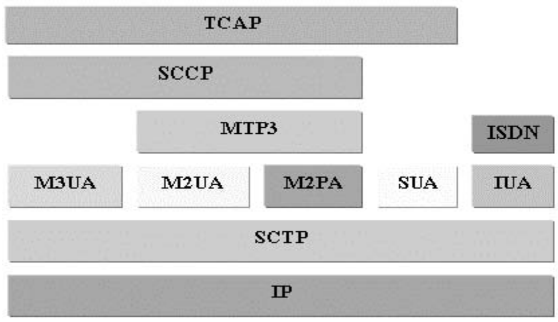

Fig 4. SIGTRAN Protocol Stack [8]

In SIGTRAN, if the native protocol is MTP (Message Transport Layer) Level 3, the SIGTRAN protocols provide the equivalent functionality of MTP Level 2. If the native protocol is ISUP or SCCP, the SIGTRAN protocols provide the same functionality as MTP Levels 2 and 3. If the native protocol is TCAP, the SIGTRAN protocols provide the functionality of SCCP (connectionless classes) & MTP Levels 2 & 3.

Presently, SS7 telephony network is connected with the IP network using media gateway controller. The existing IP telephony network comprising of very costly Media gateway controller is shown bellow [9]:

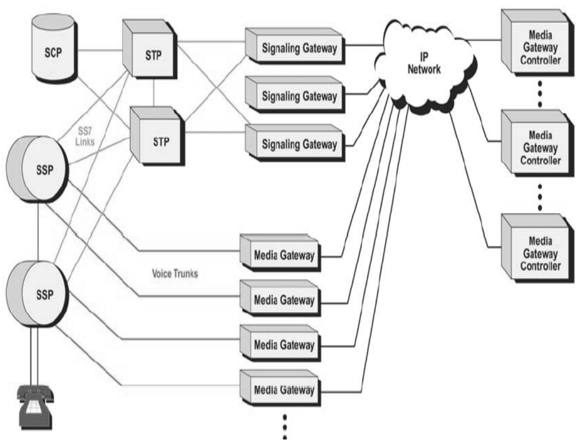

Fig 5. Existing IP telephony network structure

2. Proposed Exchange

Recently many alternative approaches are searched for alternative of costly media gateway [10]. We have proposed a CPLD driven hardware and related software which will function as a common exchange between SS7 and IP network. In our common exchange the signaling and voice data transport will take place in the following way,

II.A. Signaling data transport: SCTP will allow the reliable transfer of signaling messages between signaling endpoints in an IP network [12]. To establish an association between SCTP endpoints [13], one endpoint will provide the other endpoint with a list of its transport addresses. These transport addresses will identify the addresses that will send and receive SCTP packets.

II.B. Voice data transmission: For transmission of voice data, H.323 protocol is used [14]. Actually H.323 is not a single protocol specification, rather it's an architectural description, including references to a suite of protocols required for network integration. The protocols defined under H.323 are:

- H.323 RAS – Registration, Authentication and Status protocol
- H.225 Call Signaling
- H.245 Control Signaling
- RTP –which carries packetized media between H.323 endpoints.

Now in our proposed network architecture we have used CPLD driven device. The function of Media Gateway Controller is performed by Black Fin microprocessor. Our developed E1 card using Flacon IC and ASTERIX software installed in the Blackfin performs the task of Signaling Gateway, Media Gateway. So, our proposed system comprises only of CPLD driven device and E1 card. This makes it very cost-effective.

Our proposed network structure is shown bellow:

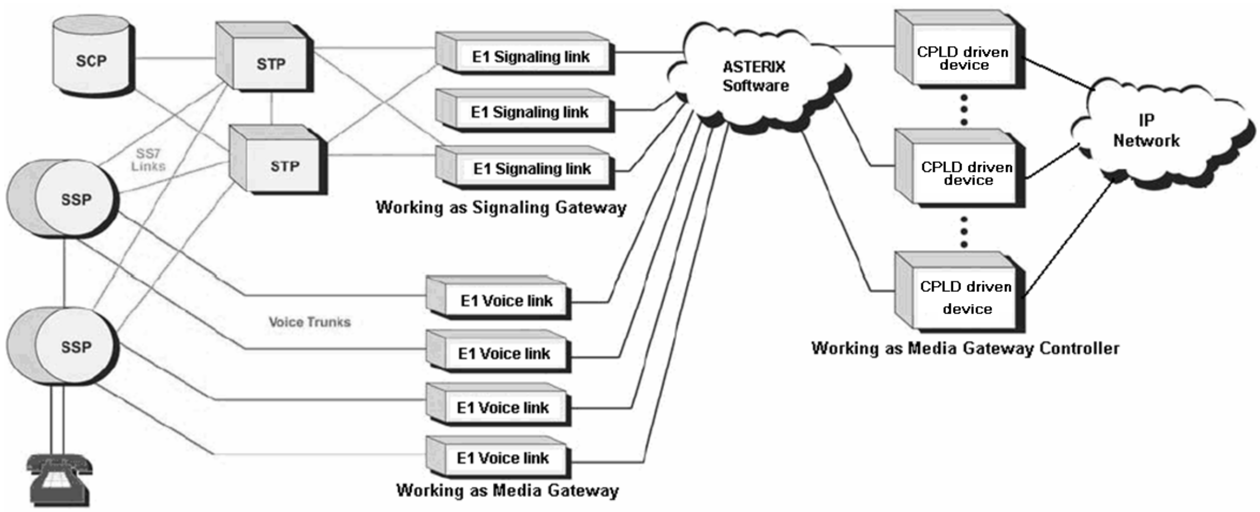

Fig 6. Proposed network structure

Using this structure many large networks can be interconnected by simple IP network. This eliminates the need of high cost circuit switching elements and sophisticated microwave link. For example, the MSC (Main Station Controller) of an SS7 network can be connected with HLR (Home Location Register) or ASG.

This remarkably reduces cost by eliminating costly existing exchanges, microwave backbone links etc. Such interconnection architecture is shown bellow:

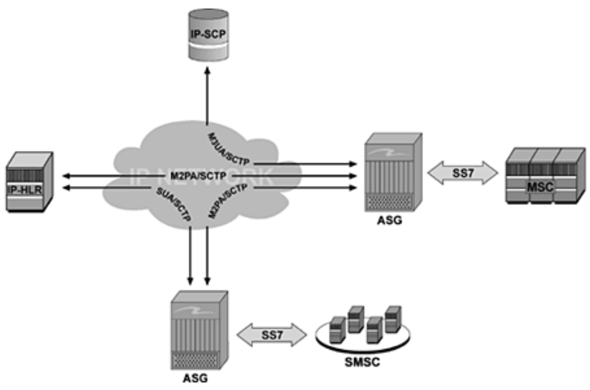

Fig 7. Interconnection of MSC, HLR and ASG using IP network.

In the SS7 links, data will be carried through SS7 protocol which has own defined structure. When we will make interconnection of two SS7 networks then data of SS7 network will be put into SIGTRAN protocol. Then these SIGTRAN protocol data will be carried by IP packets through IP network. And when IP network will terminate at SS7 network, then IP packet data will be converted to SS7 data again. This way interchanging of protocols will occur. Protocol interchanging structure will be like bellow:

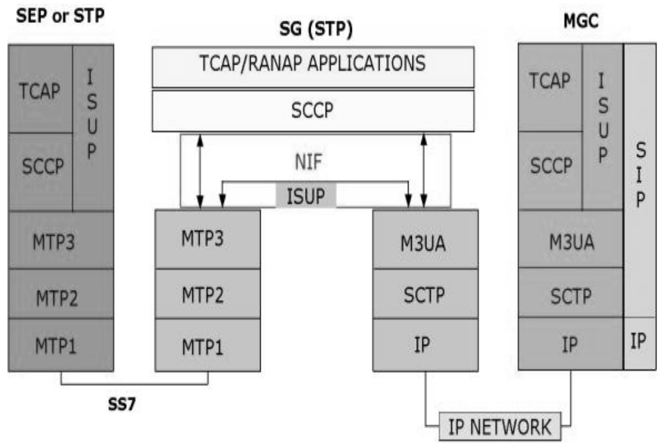

Fig 8. Protocol interchanging structure. [11]

3. A. CPLD driven board designing:

Ay first we have configured the controlling part of the device by programming CPLD. Is our case we have used XC9536XL CPLD having 34 I/O pins.
Then as processor of the device, we have chosen BlackFin microprocessor. The computational capability of the Blackfin microprocessor is enough for completing protocol interchanging task [15] [16].

The schematic diagram of the used controlling unit and processing part is shown bellow:

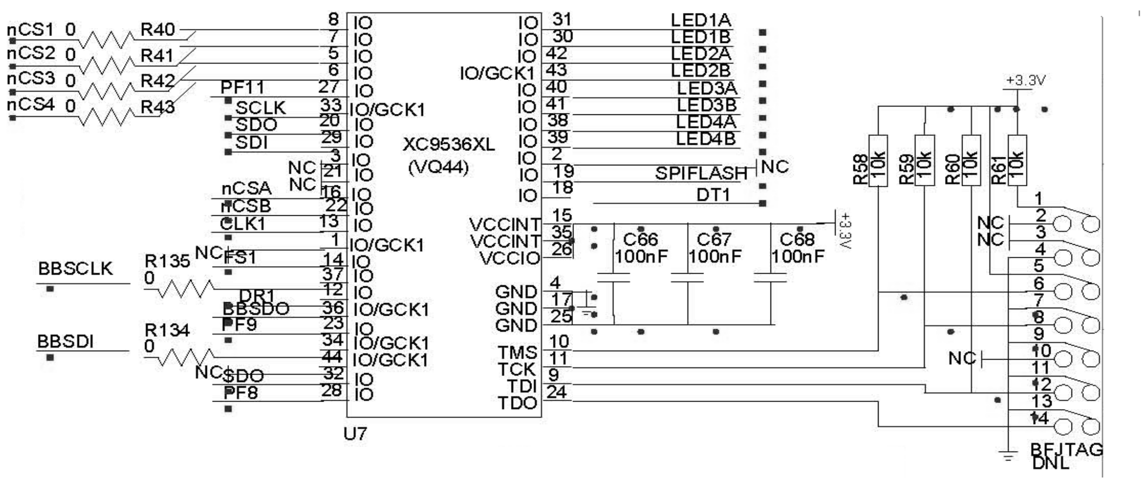

Fig 9. Schematic diagram of controlling unit using CPLD

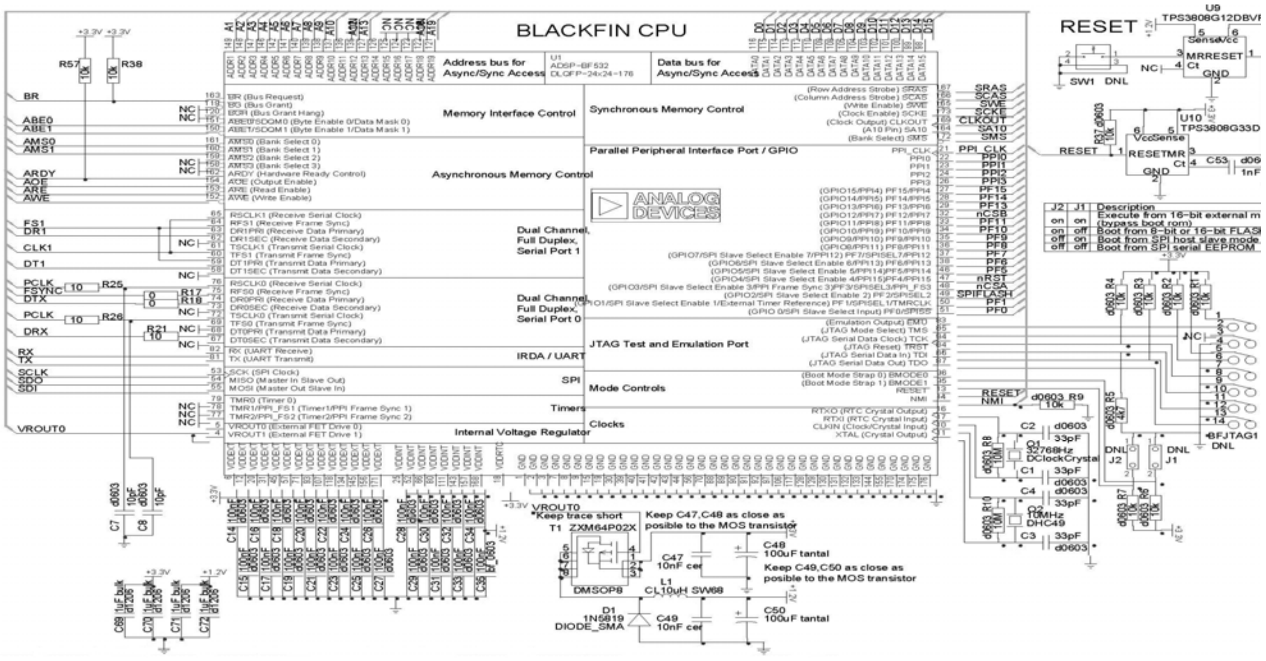

Fig 10. Schematic diagram of processing unit.

The next part includes the use of Ethernet port and 256MB NAND memory unit. Ethernet port is used for receiving and transmitting SIGTRAN data. Their used schematics are shown bellow:

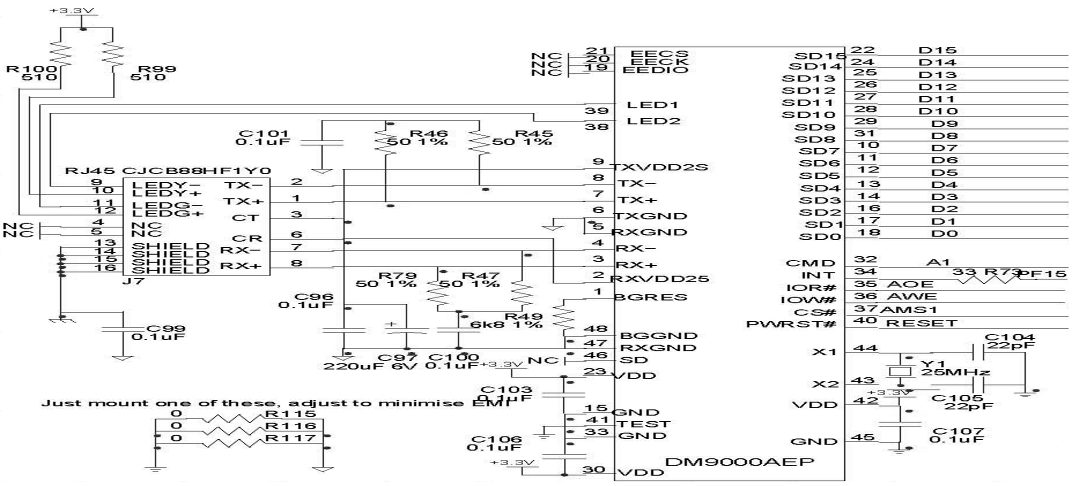

Fig 11. Schematic diagram of ETHERNET port.

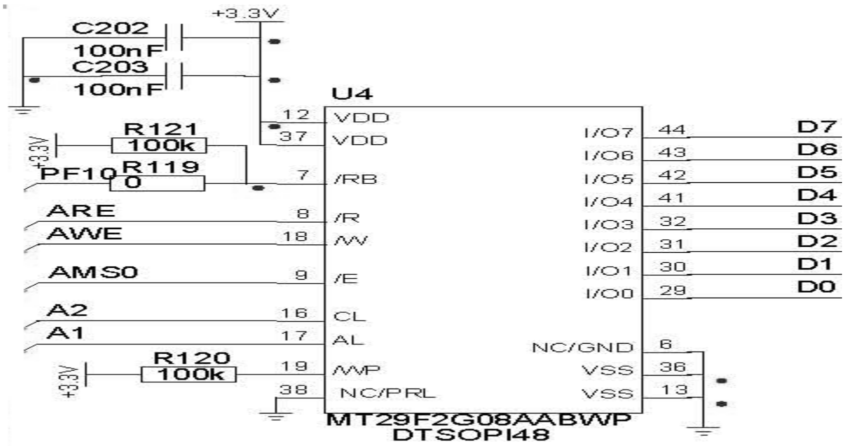

Fig 12. Schematic diagram of 256MB NAND memory unit

3. B. Modification of Asterix software:

Asterisk is a free software / open-source software implementation of a telephone private branch exchange (PBX). Like any PBX, it allows a number of attached telephones to make calls to one another, and to connect to other telephone services including the PSTN. Its name comes from the asterisk symbol '*' which in Unix (including Linux) and DOS environments represents a wildcard, matching any sequence of characters in a filename [17].

Asterisk is released under a dual license scheme, the free software under General Public License (GPL), the other being a commercial. We have used the Asterix software under GPL.

Asterisk also supports a wide range of Voice over IP protocols, including SIP and H.323. Asterisk can interoperate with most SIP telephones, acting both as registrar and as a gateway between IP phones and the PSTN. That's why we have chosen Asterix software to work with. Currently we successfully worked with E1 data, voice mail, conference calling, interactive voice response (phone menus), and automatic call distribution system of Asterix.

3. C. E1 interface designing:

For the E1 interface we have used Falcon IC. The FALC56 is the latest addition to Infineon's FALC® family of sophisticated E1/T1/J1 framer and Line Interface Unit (LIU) transceivers. The FALC56 is ideal for use in state-of-the-art wireless base stations, switches, and Internet access equipment, and is also highly suited for ISDN applications. Designed for both long and short haul applications, the FALC56 supports all standard E1/T1/J1 functions. The FALC56 comes with a wide range of support tools, designed to assist rapid hardware and software design. The schematic diagram of the used E1 card using FALC56 IC is given bellow:

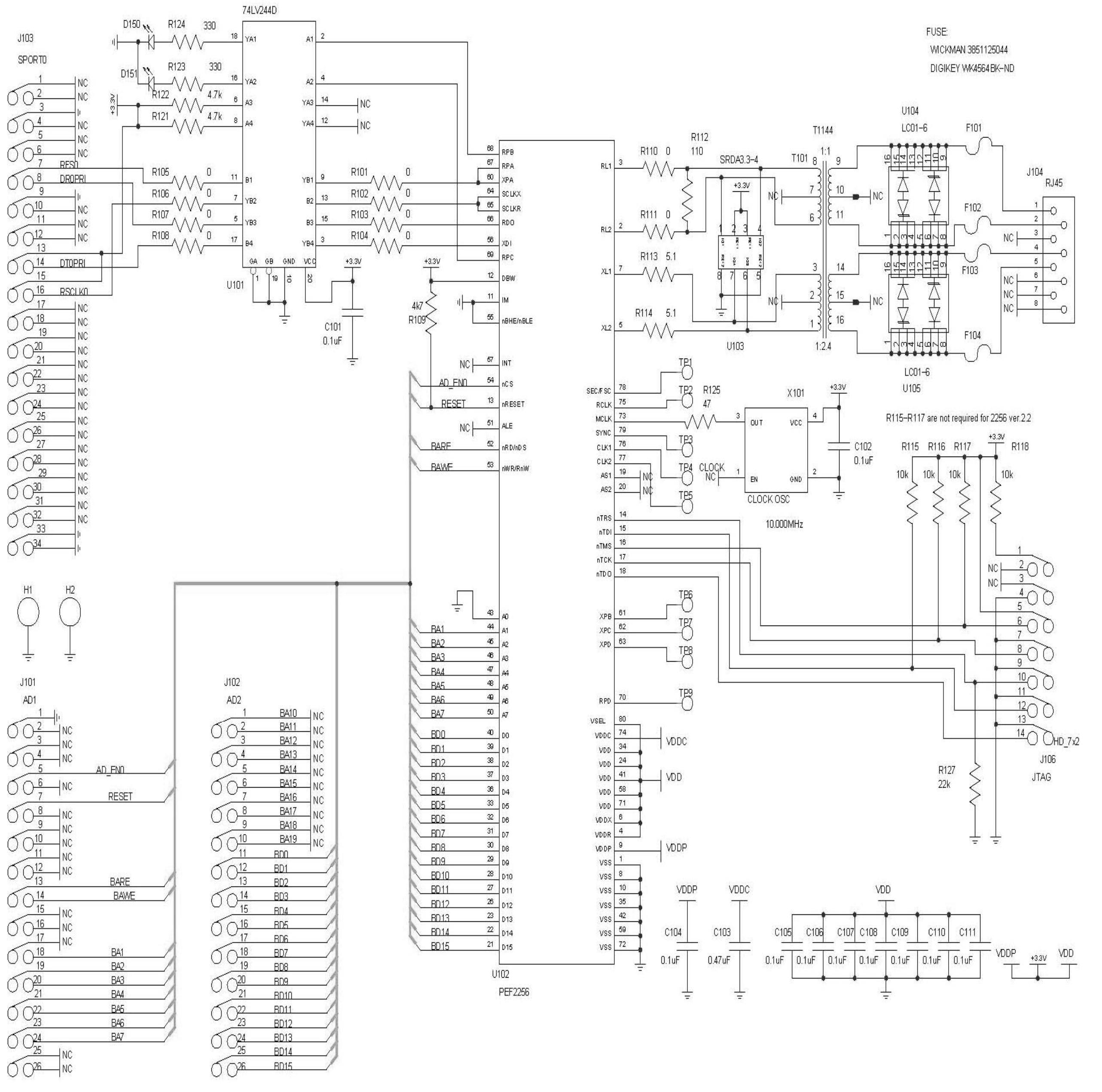

Fig 13. Schematic diagram of E1 card.

4. Capacity of the Hardware:

E1 accommodates 32 channels including 30 voices, 1 signaling and 1 synchronizing channel. Each channel requires 8KBps bandwidth (BW).
So, total BW required for 1 E1 link is = 8KBps * 32 = 2048KBps = 2MBps

Blackfin has one Fast Ethernet port with 100MBps Bandwidth.
So, total E1 link one Ethernet port can support is = 50
Here we see, up to 50 * 30 = 1500 voice channels can be supported by a Blackfin processor having a single fast Ethernet port.
And up to 50 * 1 = 50 signaling channels can be supported by a Blackfin processor having a single fast Ethernet port.

5. Outcome:

The hardware designing part is completed and currently it's working soundly. It can connect an E1 link with IP network which can replace the costly equipment used currently.
One of the major achievements of our work is the tremendous reduction of the cost. The manufacturing cost of the hardware is very minute compared to the market price of similar range hardware. Currently a single E1 card costs about 450USD whereas our hardware designing cost is only 30USD. And the development of the CPLD driven hardware costs only about 60USD. In total the whole hardware can be developed with only about 90USD.
Another important part of our work is that the whole system requires very little amount of power to operate. This will enable it to be in work at the rural area where power consumption is a vital point due to constrain in supply.

6. Conclusion:

The current demand of very high data rate compared to voice in communication network made it essential to consolidate voice and data traffic. As IP network is the most preferred solution, the media gateway is used widely which is very costly. As a very cost effective alternative solution, we have designed this common exchange for connecting existing SS7 and SIGTRAN network. As IP network is very cheap compared to SS7 network, the future expansion of a communication network can be done using IP network while keeping it connected with the SS7 network using our common exchange. Then the cost of the development of the network will be reduced significantly. Again, PC based database and VAS can be developed easily which will be connected to SS7 network via IP enabling telecom operators to provide those services to users at a very cheap rate.

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