SIP APPLICATION SERVER

GENERIC REQUIREMENTS
No. TEC/GR/SW/SAS – S01/02/ NOV. 2009

(Supersedes GR No. GR/SAS – 01/01.SEP 2007)

©TEC

TELECOMMUNICATION ENGINEERING CENTRE
KHURSHIDLAL BHAWAN, JANPATH
NEW DELHI-110 001
INDIA

GOVERNMENT OF INDIA
DEPARTMENT OF TELECOMMUNICATIONS
TELECOMMUNICATION ENGINEERING CENTRE
NEW DELHI - 110001

SIP APPLICATION SERVER

GENERIC REQUIREMENTS
No. TEC/GR/SW/SAS – S01/02/ NOV. 2009

CONTENTS

Chapter No.	Description	Page No.

History Sheet

S.No.	Title	GR No.	Remarks
1	Sip Application Server	GR/SAS – 01/01.SEP 2007	Issue 1
2	SIP Application Server New Format	No. TEC/GR/SW/SAS – S01/02/ NOV. 2009	Issue 2

CHAPTER -1

INTRODUCTION

1.1 This document specifies the Generic Requirements of SIP Application Server (SAS) that enables service providers to develop, roll out and host next – generation network (NGN) services e.g. Multimedia services, broadcasting, messaging etc.

1.2 This document describes the functional, interconnectivity and other mandatory requirements of SIP Application Server. Desirable requirements have been described in Chapter 10.

1.3 This document also describes the various types of services supported along with 3rd party ISV's (Independent Software vendors) / other application server by SIP Application Server to wire line and wireless subscribers. For all ITU-T Recommendations, ETSI, 3GPP and IETF standards/ RFC referred in this document, the latest release shall be applicable.

1.4 For all TEC documents referred in this document, the latest issue with all associated Amendments, Addendum and Corrigendum shall be applicable.

CHAPTER – 2

DESCRIPTION

2.l   Architecture

Figure 1 shows the connectivity of SIP Application Server (SAS) in NGN Architecture. SAS is one important part in the NGN Architecture. Applications providing value added services are deployed on the SAS. On request from a subscriber for a particular service, SAS is triggered by the soft switch. Soft switch will redirect certain sessions to the SAS based on the internal filters and criteria. SAS may also communicate with soft switch to get additional information about a subscriber or to be notified about changes in the profile of the subscriber and based upon this information it may provide service to soft switch subscriber. SAS shall communicate with soft switch on Session Initiation Protocol (SIP).
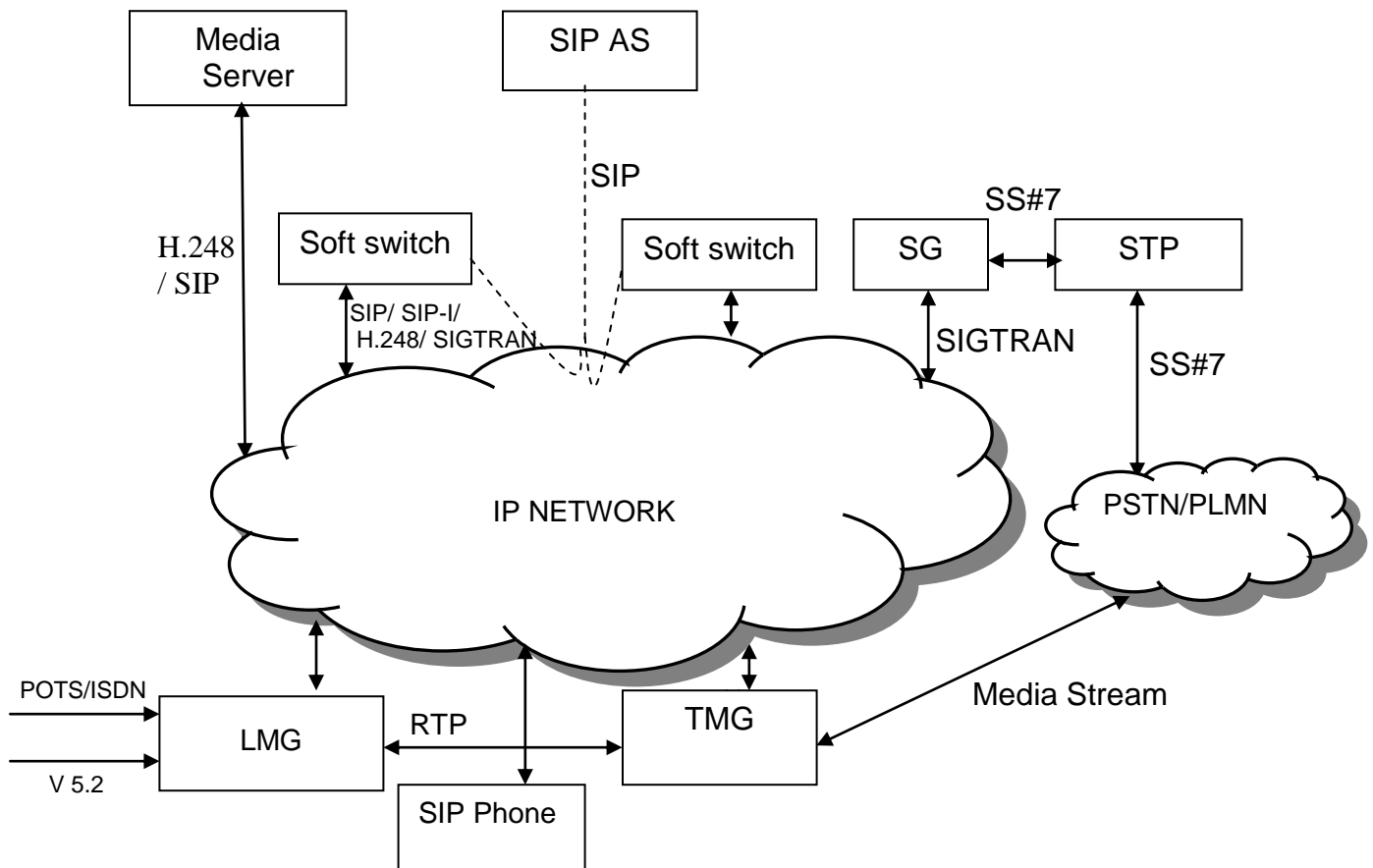
Figure - 1 Connectivity of SIP AS in NGN Architecture
(**Refer to Gr No. TEC/SW/NGN-LTS/01)**

LMG - Line Media Gateway
RTP - Real Time Protocol
SIP AS - Session Initiation Protocol Application Server
SIP-I - Session Initiation Protocol for ISUP
STP - Signaling Transfer Point
SG - Signaling Gateway
TMG - Trunk Media Gateway

One SAS may cater to many soft switches for provision of various services. One soft switch may access many SASs for the services depending upon the implementation and requirements. SAS may be located in the home network or in the external third party network.

2.2 SIP Application Server architecture shall be modular, simple to implement and able to deploy future-proof software components.

2.3 Architecture shall be component based to allow for changing of various components to obtain the desired functionality.

2.4 SIP Application Server architecture shall have single node or multi node deployment.

CHAPTER – 3

FUNCTIONAL REQUIREMENTS

3.1   General Requirements

3.1.1 SIP Application Server shall enable service providers to develop and deploy SIP based applications rapidly and cost effectively.

3.1.2 The SIP Application Server shall use fault-tolerant, highly scalable, open-ended state-of-the-art hardware, software and networking technologies and provide carrier-grade performance.

3.1.3 SIP Application Server shall have support for multi-party and multimedia call control.

3.1.4 SIP Application Server shall support Multi operating system portability.

3.1.5   SIP Application Server shall have in-built Load Monitoring to protect against overload conditions.

3.1.6   SIP Application Server shall support Event criteria filtering mechanisms.

3.1.7   SIP Application Server shall support Multi-lingual audio and video announcements.

3.2   Protocols and Standards

3.2.1   SIP Application Server shall comply with the following Protocols/ specifications:

- SIP: Session Initiation Protocol , RFC 3261
- Session Description Protocol, RFC 2327
- An Offer/Answer Model with the Session Description Protocol (SDP), RFC 3264
- Reliability of Provisional Responses in the Session Initiation Protocol (SIP), RFC 3262
- SIP INFO Method, RFC 2976
- Transport Protocol: TCP, UDP
- Shall support IP addressing as per version 4 (IP v4) as well as version 6 (IP v6)
- SIP servlet API as defined in JSR 116

3.2.2 In addition to above, support of following SIP RFCs (as per their references) is required:

S.No.	Name of RFC	Reference& No.
1.	Session Initiation Protocol Servlet API	JSR 116
2.	Transport Layer Security Protocol	IETF RFC 2246
3.	Session Description Protocol	IETF RFC 2327
4.	Session Initiation Protocol (SIP)	IETF RFC 2543
5.	A DNS RR for specifying the location of services. (DNS SRV)	IETF RFC 2782
6.	URLs for Telephone Calls.	IETF RFC 2806
7.	The SIP INFO Method	IETF RFC 2976
8.	Session Initiation Protocol (SIP)	IETF RFC 3261
9.	Reliability of Provisional Responses in Session Initiation Protocol (SIP)	IETF RFC 3262
10.	Session Initiation Protocol (SIP): Locating SIP Servers	IETF RFC 3263
11.	An Offer/Answer Model with Session Description Protocol (SDP)	IETF RFC 3264
12.	Session Initiation Protocol (SIP)-Specific Event Notification	IETF RFC 3265
13.	Support for IPv6 in Session Description Protocol (SDP)	IETF RFC 3266
14.	Session Initiation Protocol up date method.	IETF RFC 3311
15.	Integration of Resource Management and Session Initiation Protocol (SIP)	IETF RFC 3312
16.	Privacy Mechanism for Session Initiation Protocol (SIP)	IETF RFC 3323
17.	The Reason Header Field for the Session Initiation Protocol (SIP)	IETF RFC 3326
18.	Internet Media Type message / sipfrag	IETF RFC 3420
19.	Session Initiation Protocol (SIP) Extension for Instant Messaging	IETF RFC 3428
20.	Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3rd-Generation Partnership Project (3GPP)	IETF RFC 3455
21.	The Session Initiation Protocol (SIP) Refer Method	IETF RFC 3515
22.	Session Description Protocol (SDP) Bandwidth Modifiers for RTP Control.	IETF RFC 3556
23.	Diameter Base Protocol	IETF RFC 3588
24.	Session Initiation Protocol (SIP) Basic Call Flow Examples	IETF RFC 3665
25.	Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)	IETF RFC 3725
26.	A Presence Event Package for the Session Initiation Protocol (SIP)	IETF RFC 3856
27.	Presence Information Data Format	IETF RFC 3863
28.	Call Processing Language (CPL): A Language for User Control of Internet Telephony Services	IETF RFC 3880
29.	Session Initiation Protocol (SIP) " Replaces" Header	IETF RFC 3891
30.	Session Initiation Protocol (SIP) "Join" Header	IETF RFC 3911

| 31. | The tel URI for Telephone Numbers | IETF RFC 3966 |
| 32. | Session Times in the Session Initiation Protocol (SIP) | IETF RFC 4028 |

3.3   Applications / Services

SIP Application Server shall provide following types of SIP based Services / Applications as per TEC Document No. SR/NSF-01/01 (Services for NGN Subscriber).

3.3.1 SIP Based services that may be provided through Softswitch or any SIP application Server on the SIP phone.

3.3.1.1 Web-based applications such as web-based Conference with external server

This service provides simultaneous large web conferencing user(ports) with entry level single server systems. This can be enlarged simply by new servers with no conference downtime. For large scale service deployment, different type of servers such as Front-end web server, conference server, Recording server and Streaming server are used. For small conferencing deployment all the conferencing features can be installed on a single server. Conference server is integrated with audio bridges from several third- party to offer a combined web and phone-based audio conferencing solution. Different audio types can be used in a single conference deployment, presenting a unified conference bride interface to users.

3.3.1.2 Browser based Interaction for Supplementary services with server

This service is used in deploying services available in mobile networks like short message service (SMS) to their fixed networks customers. Similar services will be introduced from a computer based environment like management of large directories, way of communication, since a customer does not need anymore to remember a large set of numbers (fixed, cell-phone, e-mail, etc.) but can contact the partner by using the name, letting the network decide which is the appropriate method for establishing the communication based on the context and presence information held in the network. With respect to information services this requires the possibility to adapt the content to the capabilities of the terminal that is used in the current situation or even more the use of several devices interacting for service, like the presentation of a video on a television set, while the mobile phone is used to as voice channel and as remote control.

3.3.1.3 Multimedia services for SIP subscribers to be provided by Softswitch or external application server based on SIP.

The system shall support multimedia communications between SIP-based and soft terminals via SBC to provide multimedia service. Softswitch shall support multiple audio/video coding formats such as G.711, G.723, H.261, H.263 and H.264.

Point to point multimedia communication

The point to point multimedia communication services to be provided are as follows:

i)   **Instant messaging**: Allows real-time communication by means of text between one terminal user and another who has already logged in.

ii)   **Video communication**: With respective cameras installed, a terminal user can directly originate a video call to the opposite party and the appropriate video quality can be determined depending on the network bandwidth.

iii)   **Application share:** A terminal user shares an application; then the opposite party can use this shared application remotely. Application Sharing is an element of remote access, that enables two or more users to access a shared application or document from their respective computers simultaneously in real time. Generally, the shared application or document runs on a host computer and remote access to the shared content is provided to other users by the host user.

iv)   **Electronic whiteboard**: Both parties can write and draw on the same picture, for example, for discussion purposes. This is applicable to many occasions such as remote teaching and technical exchange. An Electronic white board is a computer-based, distributed, shared graphical and text working space. One useful application of Electronic white board is for distance learning.

v)   **Content release:** The contents of advertisements and media streams can be released through a multimedia terminal, and a platform can be used to selectively locate the user or to immediately release.

Following additional services may be provided

vi)   **Services built on industry standard, programmable platform:**

*   Service that could be easily customizable and extensible through service creation environment
*   Easy integration with external platforms
*   Portable across app servers

vii)   **End-User services:**

*   Intelligent routing
*   Voice mail routing

viii) **System/Network services**:

- PSTN Gateway Selection
- PSTN Gateway Failover
- Local Number Routing
- Address Translation

3.3.2 Other Services for SIP Subscribers.

It shall be possible to provision interactive, non-interactive and both interactive and non-interactive services to SIP and IP endpoints, as below:

Real - time Conversational Voice services. (Interoperable with the existing public- switched telephone network (PSTN) and with mobile networks).

(i) Point-to point Voice

(ii) Point-to point FAX

(iii) Multi-point voice & FAX

(iv) Click-to –dial

(v) Auto call back

CHAPTER 4

INTERCONNECTIVITY AND INTEROPERABILITY REQUIREMENTS

4.1 Application Programming Interfaces (API's):

4.1.1 SIP Application Server shall provide different Service Capability Features in the form of APIs to gain access to network capabilities, to enable flexible development of third party solutions for career-grade converged voice/data/multi-media next-generation applications.

4.1.2 SIP Application Server shall support open application interfaces based on SIP along with the support of JAIN(API related to SIP), Parley(API related to 3rd Party servers), Parlay-X(Version 2.1 or 3.0 for Web services), XML(Version 1.0 from W3C) and SOAP(Version 1.2 of W3C) APIs to enable flexible development of third party applications. Comprehensive information in the form of soft and hard copies concerning the API shall be supplied.

4.1.3 SIP servlet API as defined in JSR 116 shall be provided.

4.1.4 APIs shall be independent of access technology and independent of service capabilities in the network.

4.1.5 APIs for Services based on Capability Types

It shall be possible to implement and provision services, based on following capability types, for which APIs shall be provided:

(i) Identification

(ii) Authentication

(iii) Authorization

(iv) Location

(v) Presence and Availability

(vi) Group management (CUG/VPN)

(vii) Call Control :

- Generic

- Multi-Party

- Multimedia

- Conference

(viii) Session Control

(ix) Service Subscription (Registration) Management

(x) User and Terminal Profile Management

(xi) Generic messaging

(xii) Multimedia messaging

(xiii) Push

(xiv) Content based charging

(xv) Emergency Communication Management

(xvi) Address list management

(xvii) Prioritized communication /traffic handling

(xviii) Service Independence - service capabilities which will allow IP multimedia applications to be deployed in a vender independent manner.

(xix) Application Service inter working - Allows inter working of application service protocols. This capability should allow inter working between application services and network entities for execution of the services

4.2 SIP Application Server shall inter work with soft switch on SIP protocol and host and execute services for its subscribers.

4.3 SIP Application Server shall support web services interface and shall comply to Service Oriented Architecture (SOA).

CHAPTER - 5

QUALITY REQUIREMENTS

5.1   Quality Requirement (QR):
5.1.1 The system shall comply all the quality requirements as required for IP based systems.

5.2   Operational Requirement (OR): The system shall meet the following maintenance & operational requirements:

(i)   The design of the equipment shall not allow plugging of a module in the wrong slot or upside down.

(ii)   The removal or addition of any interface cards shall not disrupt traffic on other cards.

(iii)   All critical modules shall be identified and shall be provided in full redundant configuration.

(iv)   The equipment shall have a redundant processor configuration working in active and hot standby mode. Each processor shall have its own hard-disk for main storage. Adequate capacity for the purpose of storage shall be provided. The supplier shall provide such details for the calculation of disk capacity.

(v)   Suitable Visual indication shall be provided for displaying healthy, unhealthy operation conditions.

(vi)   A single point failure on the equipment shall not result in network or network management system downtime.

(vii)   In the event of a bug found in the software, the manufacturer shall provide patches and firmware replacement if involved, free of cost. Compatibility of the existing hardware shall be maintained with future software/firmware.

(viii) The normal operation of SIP Application Server shall not be affected while undertaking software updates, enhancement of services or correction to programs or functional blocks.

(ix)   A user-friendly Graphical User Interface (GUI) workstation shall be provided for interaction with SIP Application Server.

(x)   Test programs shall include fault tracing for detection and localization of system faults.

(xi)   Facilities shall be in-built to ensure automatic system reconfiguration on detection of any major software fault.

(xii)   In the event of a full system failure, a trace area shall be maintained in non-volatile memory for analysis and problem resolution.

(xiii) A power down condition shall not cause loss of connection configuration data storage.

(xiv) Live Insertion and hot swap of modules shall be possible to ensure maximum network availability and easy maintainability.

(xv) The hardware and software components shall not pose any problems in the normal functioning of all network elements wherever interfacing with Service Provider's network for voice, data and transmission systems, as the case may be.

(xvi) The equipment shall have availability figure of 99.999%. The MTBF (Mean Time Between Failure) and MTTR (Mean Time To Restore) predicted and observed values shall be furnished along with calculations by the manufacturer.

5.2 Element Management System (eMS) For SIP Application Server

5.2.1 The eMS can be co-located (on a LAN environment) or remotely linked (through dial up or leased line with appropriate modem). The latest hardware with scalable function of requirements shall be used.

5.2.2 The eMS shall be capable of performing all functions related to Administration, Management, Supervision and Maintenance of all kinds in real time and post processing mode. When a remote control centre is established, it shall also be possible to carry out these operations by issuing commands from the remote centre. Different categories of alarm shall be supported by SIP Application Server.

5.3 It shall be possible to copy the dump to back-up storage media and take a printout also.

CHAPTER 6

EMI/EMC REQUIREMENTS

6.1   Electromagnetic Interference

The equipment shall conform to the following EMC requirements for Class A:

General Electromagnetic Compatibility (EMC) Requirements: - The equipment shall conform to the EMC requirements as per the following standards and limits indicated therein. A test certificate and test report shall be furnished from a test agency.

a)   Conducted   and   radiated   emission *(applicable   to   telecom equipment)***:**

Name of EMC Standard: "CISPR 22 (2005) with amendment 1 (2005) & amendment 2 (2006) - Limits and methods of measurement of radio disturbance characteristics of Information Technology Equipment".

Limits:-
i) To comply with Class A or B (to be mentioned in the GR / IR as per the specific requirement) of CISPR 22 (2005) with amendment 1 (2005) & amendment 2 (2006).

ii) The values of limits shall be as per TEC Standard No. TEC/EMI/TEL-001/01/FEB-09**.**

<center>OR</center>

Conducted and radiated emission (applicable to instruments such as power meter, frequency counter etc.):
Name of EMC Standard: "CISPR 11 {2004}- Industrial, scientific and medical (ISM) radio- frequency equipment-Electromagnetic disturbance characteristics- Limits and methods of measurement"
Limits :-
i) To comply with the category of Group 1 of Class A of CISPR 11 {2004}

ii) The values of limits shall be as per clause No. 8.5.2 of TEC Standard No. TEC/EMI/TEL-001/01/FEB-09**.**

b)   Immunity to Electrostatic discharge:
Name of EMC Standard: IEC 61000-4-2 {2001} "Testing and measurement techniques of Electrostatic discharge immunity test".

Limits: -
i) Contact discharge level 2 {± 4 kV} or higher voltage;

ii) Air discharge level 3 {± 8 kV} or higher voltage;

c) Immunity to radiated RF:

Name of EMC Standard: IEC 61000-4-3 (2006) "Testing and measurement techniques-Radiated RF Electromagnetic Field Immunity test"

 Limits:-

For Telecom Equipment and Telecom Terminal Equipment with Voice interface (s)
i) Under Test level 2 {Test field strength of 3 V/m} for general purposes in frequency range 80 MHz to 1000 MHz and

ii) Under test level 3 (10 V/m) for protection against digital radio telephones and other RF devices in frequency ranges 800 MHz to 960 MHz and 1.4 GHz to 6.0 GHz.

For Telecom Terminal Equipment without Voice interface (s)
Under Test level 2 {Test field strength of 3 V/m} for general purposes in frequency range 80 MHz to 1000 MHz and for protection against digital radio telephones and other RF devices in frequency ranges 800 MHz to 960 MHz and 1.4 GHz to 6.0 GHz.

d)   Immunity   to   fast transients (burst):

Name of EMC Standard: IEC 61000- 4- 4 {2004}   "Testing and measurement techniques of electrical fast transients/burst immunity test"
Limits:-
 Test Level 2 i.e. a) 1 kV for AC/DC power lines; b) 0. 5 kV for signal / control / data / telecom lines;

e) Immunity to surges:

 Name of EMC Standard: IEC 61000-4-5 (2005) "Testing & Measurement techniques for Surge immunity test"

Limits:-
i) For mains power input ports : (a)1.0 kV peak open circuit voltage for line to ground coupling (b) 0.5 kV peak open circuit voltage for line to line coupling

ii) For telecom ports : (a) 0.5 kV peak open circuit voltage for line to ground (b) 0.5 KV peak open circuit voltage for line to line coupling.

f) Immunity to conducted disturbance induced by Radio frequency fields:

Name of EMC Standard: IEC 61000-4-6 (2003) with amendment 1 (2004) & amd. 2 (2006) "Testing & measurement techniques-Immunity to conducted disturbances induced by radio- frequency fields "

 Limits:-
 Under the test level 2 {3 V r.m.s.}in the frequency range 150 kHz-80 MHz for
 AC / DC lines   and Signal /Control/telecom lines.

g)   Immunity to voltage dips & short interruptions (applicable to only ac mains power input ports, if any):

Name of EMC Standard: IEC 61000-4-11 (2004) "Testing & measurement techniques- voltage dips, short interruptions and voltage variations immunity tests"

Limits:-

i) a voltage dip corresponding to a reduction of the supply voltage of 30% for 500ms(i.e. 70 % supply voltage for 500 ms)

ii) a voltage dip corresponding to a reduction of the supply voltage of 60% for 200ms; (i.e. 40% supply voltage for 200ms) and

iii) a voltage interruption corresponding to a reduction of supply voltage of > 95% for 5s.

Note 1 : Classification of the equipment:

Class B: Class B is a category of apparatus which satisfies the class B disturbance   limits. Class B is intended primarily for use in the domestic environment and may include:

- Equipment with no fixed place of use; for example, portable equipment powered by built in batteries;
- Telecommunication   terminal   equipment   powered   by   the telecommunication networks
- Personal computers and auxiliary connected equipment.

Please note that the domestic environment is an environment where the use of broadcast radio and television receivers may be expected within a distance of 10 m of the apparatus connected.

Class A: Class A is a category of all other equipment, which satisfies the class A limits but not the class B limits.

Note 2: The test agency for EMC tests shall be an accredited agency and details of accreditation shall be submitted.
Alternatively EMC test report from a non-accredited test lab, which is audited by an accredited lab / accrediting authority for the availability of all the essential facilities (test equipment, test chamber, calibrations in order, test instructions, skilled personnel etc.), required for performing the tests according to the EMC test methods audited, may be acceptable.

However, such accredited lab / accrediting authority should take responsibility of the test results of the "non accredited lab" along with indication of period of such delegation and the submitted test report should be of such valid period of delegation. The audit report, mentioning above facts, should be provided along with EMC test report.

Note 3 :- For checking compliance with the above EMC requirements, the method of measurements shall be in accordance with TEC Standard No. TEC/EMI/TEL-001/01/FEB-09 and the references mentioned therein unless otherwise specified specifically. Alternatively, corresponding relevant Euro Norms of the above IEC/CISPR standards are also acceptable subject to the condition that frequency range and test level are met as per above mentioned sub clauses (a) to (g) and TEC Standard No. TEC/EMI/TEL-001/01/FEB-09. The details of IEC/CISPR and their corresponding Euro Norms are as follows:

IEC/CISPR	**Euro Norm**
CISPR 11	EN 55011
CISPR 22	EN 55022
IEC 61000-4-2	EN 61000-4-2
IEC 61000-4-3	EN 61000-4-3
IEC 61000-4-4	EN 61000-4-4
IEC 61000-4-5	EN 61000-4-5
IEC 61000-4-6	EN 61000-4-6
IEC 61000-4-11	EN 61000-4-11

CHAPTER - 7

SAFETY REQUIREMENTS

7.1   Safety Requirements

7.1.1 The operating personnel shall be protected against shock hazards as per IS 8437(1993)-"Guide on the effects of current passing through the human body" [equivalent to IEC publication 60479-1(1984)]. The manufacturer/supplier shall submit a certificate in respect of compliance to these requirements

7.1.2 The equipment shall conform to IS 13252 (2003)-"Safety of information technology equipment including electrical business equipment" [equivalent to IEC publication 60950{2001}] and IS 10437 {1986} "Safety requirements of radio transmitting equipments" [equivalent to IEC publication 60215]. The manufacturer/supplier shall submit a certificate in respect of compliance to these requirements.

CHAPTER – 8

SECURITY REQUIREMENTS

8.1   Suitable   safe guards shall be provided in the man-machine communication programs to debar unauthorized persons from making any changes in the data contents stored in memory.

8.2   The man-machine language shall have facility for restricting the use of certain commands or procedures to certain staff/terminals.

8.3   **Requirement Specific to leap year Problem**

The system hardware/software shall not pose any problem due to change in date and time by events such as changeover of leap year etc., in normal functioning of the system.

8.4   Facility shall be provided for lawful Interception & Monitoring at SIP Application Server as per TEC document GR No.GR/LIS-01 for all the services provided by SAS.

CHAPTER – 9

OTHER MANDATORY REQUIREMENTS

9.1   SIP Application Server shall support numbering and IP addressing as per the 'National Numbering Plan'.

9.2   **Power Supply**

Option 1:

The equipment shall be capable of working with –40 V to –57 V. D .C. input from power supply.

Switching Mode Power Supply (SMPS) and VRLA battery to be used shall be as per TEC Generic Requirements No.GR/SMP-01/05.JAN 2005 and GR/BAT-01/03.MAR 2004 respectively. Power supply and battery shall be modular and expendable to support the ultimate equipment configuration.

Option 2:

AC Mains supply of 220 Volts with a tolerance of -15% to + 10% would be available. The frequency may be 50 Hz $\pm$ 2 Hz.   UPS and other power requirements are to be specified by the system developer. Relevant TEC Specification/ Generic Requirements as applicable may be referred

CHAPTER - 10

DESIRABLE REQUIREMENTS

(Operator Specific Requirements)

10.0 This chapter describes the desirable requirements for SIP Application Server. Clause 10.1 describes the Operator specific requirements for IMS interworking only. It may be required whenever it is deployed in future. In that case, in addition to the compliance of clause 3.2 it shall also comply with the functionalities and standards as mentioned in clause no. 10.1. Clause no. 10.2 (clause No. 10.2.1 to 10.2.13) gives the general desirable requirements.

10.1 Requirements for SIP Application Server to interwork with IMS

10.1.1 It shall be possible for SIP Application Server to inter work with IP Multimedia Subsystem (IMS) and provide various services to its subscribers also. Figure - 2 shows the connectivity of SIP Application Server in the IMS Architecture.

The SIP AS is triggered by the S-CSCF which will redirect certain sessions to the SIP AS based on the internal filters and criteria or by requesting filter information from the HSS. SIP AS may also communicate with HSS to get additional information about a subscriber or to be notified about changes in the profile of the subscriber.
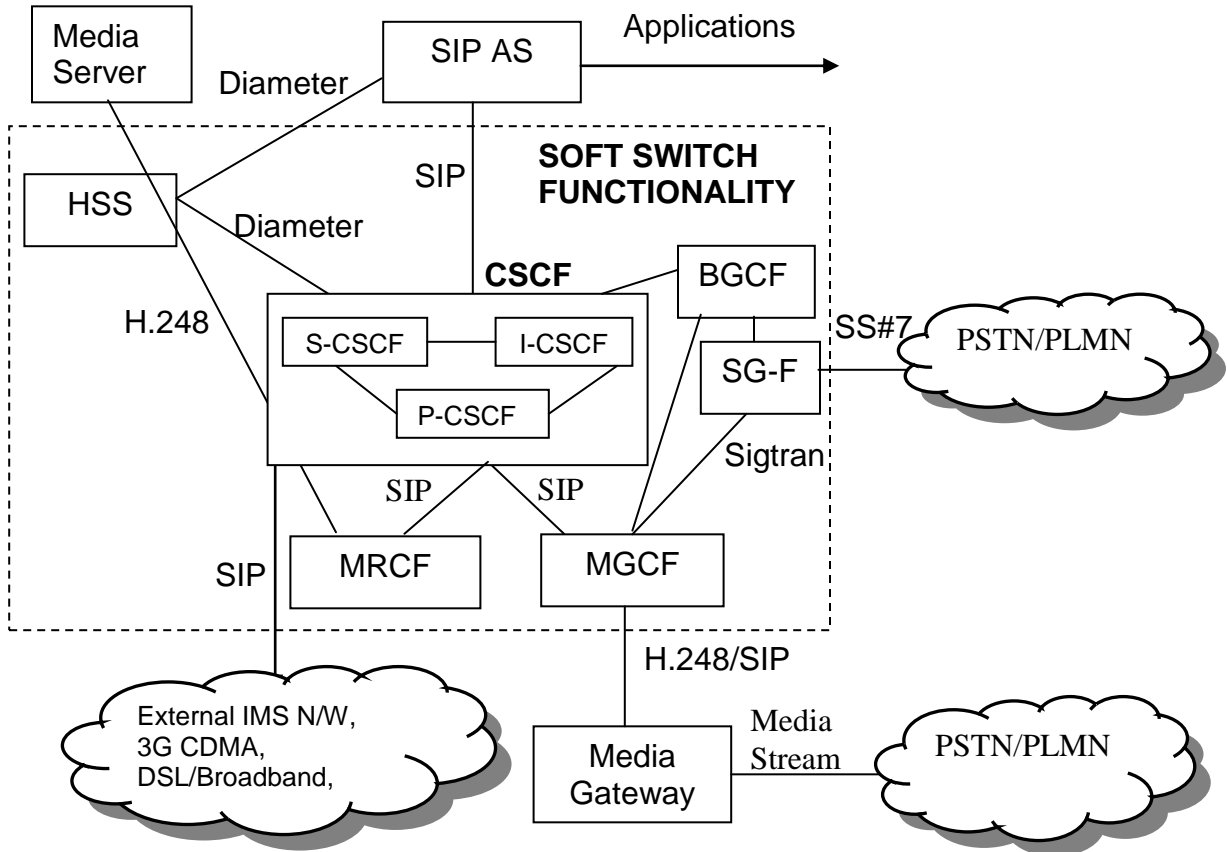

Figure - 2 Connectivity of SIP AS in IMS Architecture

SIP AS – Session Initiation Protocol Application Server
BGCF -   Breakout Gateway Control Function
CSCF - Call Session Control Function (S – Serving, P – Proxy, I – Interrogating)
MGCF - Media Gateway Control Function
MRCF - Media Resource Control Function
HSS   - Home Subscriber Server
SG-F   - Signaling Gateway Function

10.1.2 It shall be possible for SIP Application Server or via additional application server to have the following additional functionalities to inter work with IMS. Figure – 3 shows the Functional Architecture for support of service provision for IMS.

   i)   IM-SSF (IP Multimedia Service Switching Function): for CAMEL (Customized Application for Mobile Network Enabled Logic) applications using CAP as specified in 3GPP TS 29.078.

   ii)   OSA SCS (Open Service Access - Service Capability Server): to provide way for third party secure access to the IMS. It interfaces with the OSA Application Server using OSA API as specified in 3GPP TS 29.198.

   iii)   SCIM (Service Capability Interaction Manager): to perform the role of interaction management between other application servers.
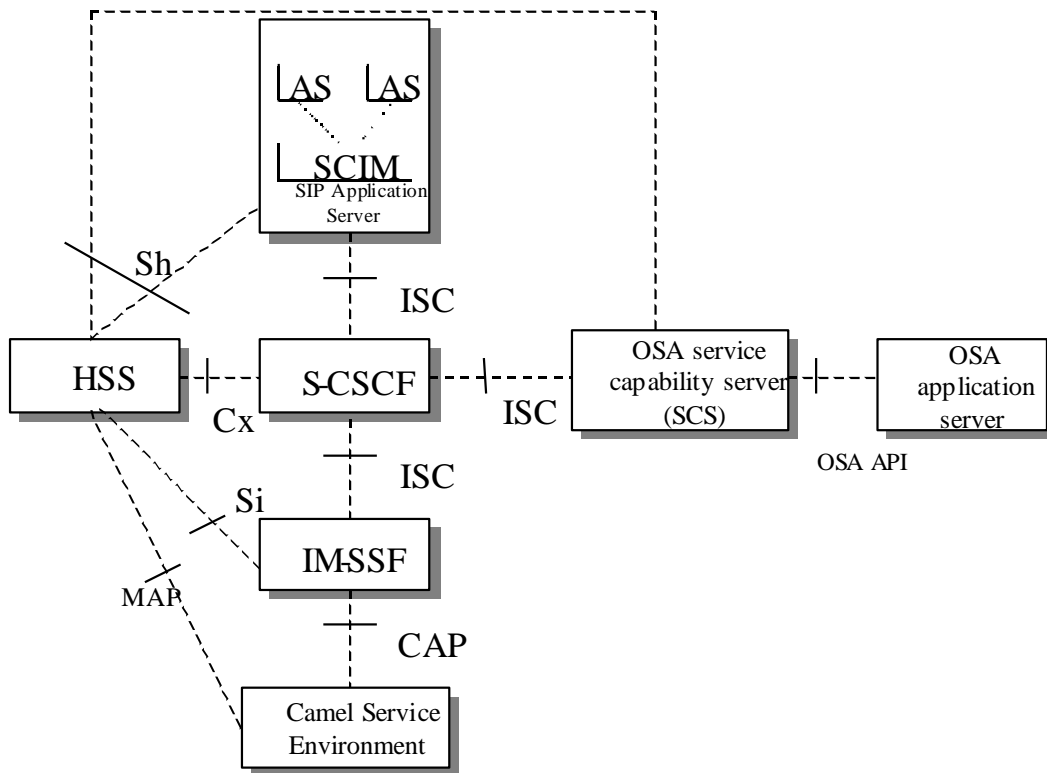

Figure-3   Functional architecture for the provision of service in the IMS

CAMEL - Customised Application for Mobile network Enhanced Logic.
IM-SSF - IP Multimedia Service Switching Function
HSS - Home Subscriber Server
OSA - Open Service Access
S-CSCF - Serving Call Session Control Function
SCIM - Service Capability Interaction Manager

10.1.3 It shall inter work with IMS components on standard Interfaces and protocols as described below: Figure-3 above shows the standard Interfaces.

i) IMS Service Control (ISC) Interface: Based on SIP (3GPP TS 23.228, TS 24.229) between Application Server (SIP AS/ IM-SSF/ OSA-SCS) and S-CSCF.

ii) Sh (Diameter) Interface: (3GPP TS 29.328, TS 29.329) between SIP Application Server/OSA SCS and HSS. HSS serves for a common subscriber database to be used for applications deployed on Application Server.

iii) Si (Diameter) Interface: (3GPP TS 23.228) between IM-SSF and HSS Home Subscriber Server).

iv) Dh (Diameter) Interface: (3GPP TS 23.228) between Application Server (SIP AS/ IM-SSF/ OSA-SCS) and SLF (Subscriber Location Function). SLF is used to select the proper HSS where there are more than one HSS.

v) Ut Interface: Between UE (User Equipment) and SIP Application Server.

vi) Ro/Rf (On line/Off line charging) Interface: Based on Diameter (3GPP TS 32.225,TS 32.299), Ro is Online Charging Reference point between SIP AS and OCS (Online Charging System) and Rf is Offline Charging Reference point between SIP AS and CDF(Charging Data function).

10.1.4 Open Service Access (OSA) Compliance:

SAS shall comply with at least following OSA standards:

- 3GPP TS 29.198-04-3-600 OSA API Part4-SubPart3-MPCC SCF
- 3GPP TS 29.198-04-3-600 OSA API Part4-SubPart4-MMCC SCF
- 3GPP TS 29.998-04-4-500 Part4-SubPart4-MPCC ISC

10.2 General desirable requirements :

10.2.1 Various dimensioning parameters e.g. traffic handling capacity, storage type and volume shall be indicated.

10.2.2 **Environment Conditions**: The purchaser shall specify the requirements of Environment Conditions as per their manual and relevant category of equipment. to be met by SAS.

10.2.3 The details for the following are to be furnished by the purchaser

- Installation
- Field Trial

- Validation
- Acceptance Testing
- Software Maintenance

10.2.4 Peak Load of processor.

10.2.5 Capacity of SIP Application Server.

10.2.6 Details of hardware, software and number of operator terminals for operation, administration and maintenance.

10.2.7 The MTBF (Mean Time Between Failure) and MTTR (Mean Time To Restore) predicted and observed values shall be furnished along with calculations by the supplier.

10.2.8 The system shall comply with quality standards like ISO 9002 or ISO 9001: 2000 certification.

10.2.9 The period for which the maintenance spares are required, may be specified by the tendering authority.

10.2.10 Number of copies (hard and soft) of following documents required shall be specified by the purchaser.

i). System description documents

ii). System operation and maintenance documents

iii). Training documents

iv). Installation Documents

v). Repair related documents

Further details of documentation required have to be specified by the purchaser.

10.2.11 List of existing equipment required to inter-work.

10.2.12 Tools and Testers.

10.2.12 Purchaser shall specify the services to be provisioned through SIP Application Server and number and kind of the user terminals.

10.2.13 Purchaser may specify the power requirement as per option 1 or 2 of clause 9.3.

Glossary:

3GPP	3rd Generation Partnership Project
AGW	Access Gateway
AN	Access Network
AS	Application Server
BGCF	Breakout Gateway Control Function
BHCA	Busy Hour Call Attempt
CAP	Camel Application Part
CAMEL	Customised Application for Mobile Network Enhanced Logic.
CDMA	Code Division Multiple Access
CISPR	Comité international spécial des perturbations radio-électriques (Special International Committee on Radio Interference)
CSCF	Call Session Control Function
DOT	Department of Telecommunication
DTMF	Dual Tone Multi Frequency
EMC	Electro Magnetic Compatibility
eMS	Element Management System
GR	Generic Requirements
GSM	Global System For Mobile Communication
GUI	Graphical User Interface
HSS	Home Subscriber Server
IEC	International Engineering Consortium
I-CSCF	Interrogating-CSCF
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IM-SSF	IP Multimedia Service Switching Function
ISDN	Integrated Services Digital Network
ISV	Independent Software Vendors

IP	Internet Protocol
ITU	International Telecommunication Union
IVRS	Interactive Voice Response System
JAIN	Java Advanced Intelligent Network
JSR	Java Specification Request
LMG	Line Media Gateway
MAP	Mobile Application Part
MGCF	Media Gateway Control Function
MRFC	Media Resource Function Controller
MRFP	Media Resource Function Processor
MS	Media Server
MTTR	Mean Time to Restore
MGC	Media Gateway Controller
NGN	Next Generation Network
NMS	Network Management System
OMC	Operation and Maintenance Centre
OSA	Open Service Access
PC	Personal Computer
P-CSCF	Proxy-CSCF
POTS	Plain Old Telephone System
PSTN	Public Service Telecom Network
QA	Quality Assurance
RF	Radio Frequency
RFC	Request for comment
RTP	Real-time Transport Protocol
SCIM	Service Capability Interaction Manager
S-CSCF	Serving-CSCF
SCS	Service Capability Server
SDP	Session Description Protocol
SG	Signaling Gateway
SIP	Session Initiation Protocol
SLF	Subscriber Location Function

SNMP	Simple Network Management Protocol
SOAP	Simple Object Access Protocol
TEC	Telecom Engineering Centre
TMG	Trunk Media Gateway
UE	User Equipment
XML	Extensible Markup Language

End of document