Name of Course : E1-E2 CFA

Chapter 7A

Topic : SIP

Date of Creation : 28.03.2011

Session Initiation Protocol

Introduction

SIP (Session Initiation Protocol) is a signaling protocol used to create, manage and terminate sessions in an IP based network. . Session Initiation Protocol (SIP) is an application-layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants. These sessions include Internet telephone calls, multimedia distribution, and multimedia conferences.SIP has been the choice for services related to Voice over IP in the recent past. It is a standard (RFC 3261) put forward by Internet Engineering Task Force (IETF). It SIP is still growing and being modified to take into account all relevant features as the technology expands and evolves. But it should be noted that the job of SIP is limited to only the setup and control of sessions. The details of the data exchange within a session e.g. the encoding or codec related to an audio/video media is not controlled by SIP and is taken care of by other protocols.

A Brief History of SIP

Initially only the traditional switch-based telephone system was the main medium for transmitting messages. However with advent of the Internet, need was felt to fabricate a system, which connects people over the IP based network. Different communities put forward different solutions but the solution presented by IETF was finally accepted as most general one.

February 1996 Initial Internet drafts were produced in the form of - Session Invitation Protocol (SIP), Simple Conference Invitation Protocol (SCIP). SIP was originally intended to create a mechanism for inviting people to large-scale multipoint conferences on the Internet Multicast Backbone (Mbone). At this stage, IP telephony didn't really exist. The first draft was known as "draft-ietf-mmusic-sip-00”. It included only one request type, which was a call setup request.

January 1999 The IETF published the draft called "draft-ietf-mmusic-sip-12". It contained the six requests that SIP has today.

March 1999 SIP published RFC 2543 as a standard. It was modified further to generate the more modern version of RFC 3261.

Functions of SIP

SIP is limited to only the setup, modification and termination of sessions. It serves Five major purposes.

- User location: determination of the end system to be used for communication;
- User availability: determination of the willingness of the called party to engage in communications;
- User capabilities: determination of the media and media parameter to be used;
- Session setup: "ringing", establishment of session parameters at both called and calling party;
- Session management: including transfer and termination of sessions, modifying session parameters, and invoking services.

Components of SIP

SIP is a signaling protocol that handles setup, modification, and termination of multimedia sessions. Though SIP messages will be transported through the same physical path as used by Media stream, SIP Signalling should be considered separately from the media. SIP messages can pass via one or more servers to find out destination address, on the other hand Media stream will be direct i.e. end to end.

SIP defines two basic classes of network entities : **client and server**. A client is any network element that sends SIP requests and receives SIP responses. Clients may or may not interact directly with a human user. A server is a network element that receives requests in order to service them and sends back responses to those requests. Four different types of servers exists: proxies, user agent servers, redirect servers, and registrars.

Proxy Servers: An intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients. A proxy server primarily plays the role of routing, which means, its job is to ensure that a request is sent to another entity "closer" to the targeted user. Proxies are also useful for enforcing policy (for example, making sure a user is allowed to make a call). A proxy interprets, and, if necessary, rewrites specific parts of a request message before forwarding it.

Redirect Servers: A redirect server is a server that accepts SIP requests, maps the destination address to zero or more new addresses and returns the translated address to the originator of the request. After that originator will contact on given addresses directly.

Registrar: A registrar is a server that accepts REGISTER requests and places the information it receives in those requests into the location service for the domain it handles.

As one machine / software can work as UAC or UAS depending upon the transaction in which it is engaged same is equally applicable for servers. Similarly, the same software can act as a proxy server for one request and as a redirect server for the next request. Proxy, location, and registrar servers defined above are logical entities; implementations may combine them into a single application.

SIP Protocol Operation

SIP is a text-based protocol, similar to HTTP (Hypertext Transfer Protocol). SIP messages are either request from a client to a server or responses from a server to client. Each message, whether a request or response, contains a start line followed by zero or more headers and is optionally followed by a message body. Message headers provide additional information regarding the request or response. The message body describes type of session to be established, including a description of media to be exchanged. It is important to mention that SIP does not define the structure or content of message body. It is defined by other protocols like SDP(Session Description Protocol). The job of SIP is to carry that description upto destination.

SIP Request: A SIP request consists of a request line, headers, an empty line and a message body. A Request-Line contains a method name, a Request-URI, and the protocol version.

The method name indicates the type of request. The core SIP specification defines six types of SIP requests, each of them with a different purpose.

The Request-URI indicates the next hop, which is where the request has to be routed. URI (Universal Resource Identifier) is an address and in the form of user@host, which is similar to an email address. It may look like : " SIP: xyz@bsnl.co.in"

Finally the protocol version of SIP is 2.0. Hence, one INVITE request may have the following format:

INVITE sip:xyz@bsnl.co.in SIP/2.0

Command	**Meanings**
INVITE	Invites a user to a call
ACK	Acknowledgement is used to facilitate reliable message exchange for INVITEs
BYE	Terminates a connection between users
CANCEL	Terminates a request, or search, for a user. It is used if a client sends an INVITE and then changes its decision to call the recipient.
OPTIONS	Solicits information about a server's capabilities.
REGISTER	Registers a user's current location
INFO	Used for mid-session signaling

SIP Response: A SIP response consists of a status line, several headers, an empty line and a message body. The message body is optional; some responses do not carry it.
Status Line: A status line has three elements: protocol version, status code and a reason phrase. The current protocol version is written as SIP/2.0. The status code reports transaction status. Status codes are integers from 100 to 699 and are grouped into six different classes. The reason phrase is meant for human eyes only. It is not meant for computers processing SIP response.

SIP/2.0 180 Ringing.

Table shows the response group, their descriptions and examples:

Response Code	Description	Example
1xx	Informational – Request received, continuing to process request	180 Ringing 181 Call is being forwarded
2xx	Success – Action was successfully received, understood and accepted.	200 OK
3xx	Redirection – Further action needs to be taken in order to complete the request	300 Multiple choices 302 Moved temporarily
4xx	Client Error – Request contains bad syntax or cannot be fulfilled at this server	404 Not found 408 Request timeout
5xx	Server Error – Server failed to fulfill an apparently valid request	503 Service unavailable 504 Version not supported
6xx	Global failure – Request is invalid at any server	600 Busy everywhere 604 Does not Exist anywhere

SIP Headers: SIP Headers are included in a request / response in order to provide further information about the message. Depending on the request/response certain headers are mandatory, some headers are optional and some headers are not applicable. Four main categories of header exist:

General Header: General Headers can be used within both requests and responses. Examples are

To Header: It indicates recipient of the request. It is important to make a distinction between To header of a request and Request – URI. The To Header is intended for the actual destination UA. Proxies cannot change it. On the other hand the Request – URI is the address of next hop in the signaling path and is therefore changed by every proxy.

From Header: It indicates the originator of the request.

Call – ID: It uniquely identifies a specific invitation to session.

Request Headers: Request Headers apply only to SIP request and used to provide additional information to the server regarding the request itself or regarding the client.
Example: Subject header, which is used to describe a textual description of the session.

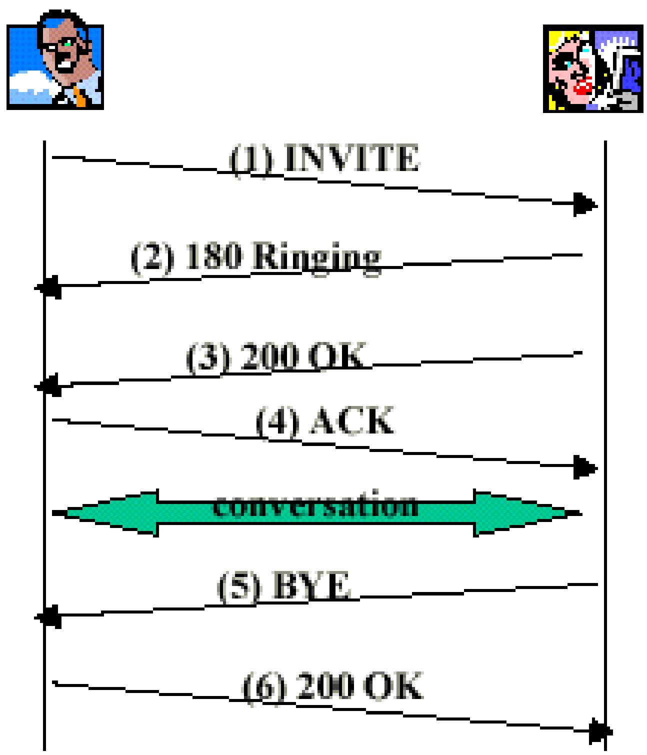
Response headers: They apply to responses only to provide further information regarding response.
Example: Retry after header, which can indicate when a called user will be available in case called user is busy or unavailable.

Entity Header: In SIP, the message body contains information about the session or information to be presented to the user. The purpose of entity header is to indicate type

and format of information included in the message body, so that appropriate application can be called upon to act on the information within the message body.
Example: Content- Length: It indicates the length of the message body in octets.

Basic SIP Operation

Basic SIP operation is the transaction of REQUESTs and RESPONSEs. Client generates SIP requests, server receives those requests and returns responses. The basic SIP operation starts with a SIP INVITE (request) message, which is generated by calling party. The message invites the called party to participate in a session i.e. a call. After reception of INVITE called party UA can generate different responses like ringing, busy, queuing, OK etc. For instance called party UA returns ringing. Subsequently, when called party answers the call, this UA will generate OK response. The calling client acknowledges that the called party has answered by issuing an ACK message. At this point, media (voice, data, video) are exchanged. Finally, one of the parties hangs up, which causes a BYE message to be sent. The other party UA sends OK response. At this point, the call is over.



Name of Course : E1-E2 CFA

Chapter 7B

Topic : IMS

Date of Creation : 29.03.2011

3 GPP IP MULTIMEDIA SUBSYSTEM (IMS)

INTRODUCTION

Operators are looking for quick and flexible ways to respond to new business opportunities. As users expand their voice telephony behaviour into more data-oriented and multimedia services, operators want to deliver a seamless and consistent user experience wherever and however the services are accessed. Also, operators need to be able to charge on the basis of value, and not just time or volume. This means it has to be easy for operators to create, implement and charge for service bundles that attract repeated use over an extended period. It must be possible to deliver these services at a cost that the user is willing to accept – and charge on the basis of value rather than time. Naturally, the services must meet user needs for ease-of-use, manageability and seamless access – there must be no barriers to accessing the desired content and services.

At the same time, operators do not want service creation and provisioning costs to spiral out of control as the service mix expands.

Both fixed and mobile operators face problem of subscriber churn, and the issue is getting worse as new service providers offers cheap, or free, calls over the Internet continue to arrive on the scene and gain market share.

- One key way to attract and retain subscribers is to offer differentiation in areas like personalization, service bundling, co-branding, business-to-business relations, tariffs, single sign-on and quality of service.
- Another key way to retain subscribers is to build on and strengthen the customer relationship so that subscribers are far more reluctant to switch suppliers, even if switching means lower call charges in the short term.

IMS offers standardized service enablers and network interfaces that will make interoperability of new MM services easier to achieve.

IMS is a tool for operators to that enable the creation and delivery of PS based person-to-person MM services in a way that protects the operator business model and generates new revenue.

Service scalability is solved by the IMS architecture. It offers support to compose services and expand existing services.

The core of IMS is combining the best of two worlds datacom industry & telecom industry.

Why IMS?

Operator perspective	End-user perspective	General
Quality Of Service	New, exciting services and enhancements of existing services	Faster time to market with new services
Service Integration	Same services available regardless of terminal and access type	Grow and protect subscriber base, increase ARPU
Keeps charging relation with user	Ease of use & Security	Controlling CAPEX and OPEX

IMS Standardization

The IMS was initially standardized by the 3rd Generation Partnership Projects (3GPP) as part of its Release 5 specifications_& is practically speaking targeted at supporting non – real time services .The second release is 3GPP Release 6 & is targeted at supporting real time services .3GPP release added inter-working with WLAN.

With the increasing penetration of Wireless Local Area Networks (WLANs) and emerging Wireless Metropolitan Area Networks (WiMax) as access network technologies, the IMS scope is now extended within the ongoing Release 7 standardization for any IP access network, including fixed access networks, i.e. DSL.

IMS Architecture as defined by 3 GPP

The IMS provides all the network entities and procedures to support real-time voice and multimedia IP applications. It uses SIP to support signaling and session control for real-time services. Fig. 1 illustrates the IMS functional architecture. The main functional entity in an IMS is the Call State Control Function (CSCF). A CSCF is a SIP server. Depending on the specific tasks performed by a CSCF, CSCFs can be divided into three different types.

- Serving CSCF (S-CSCF).
- Proxy CSCF (P-CSCF).
- Interrogating CSCF (I-CSCF).

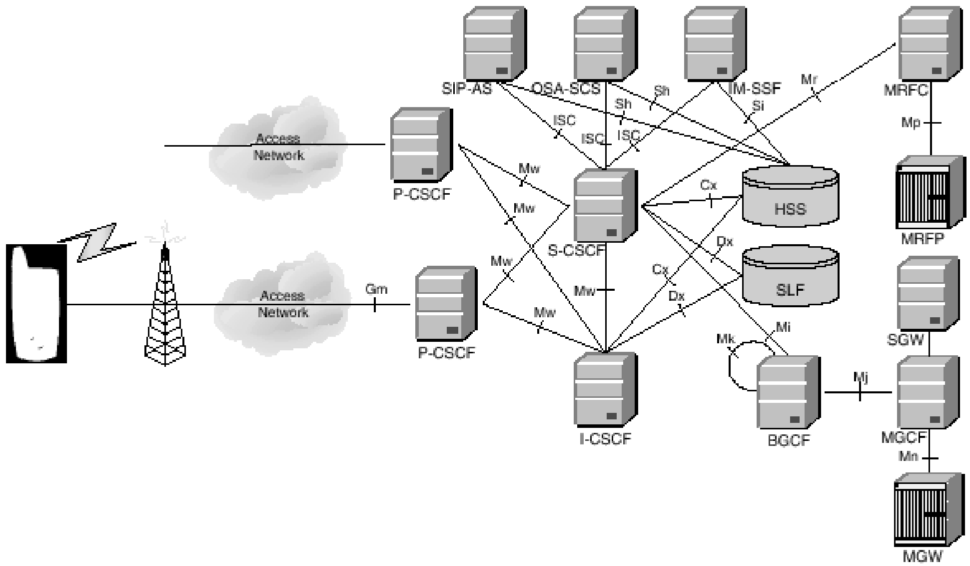


Fig. 1. 3 GPP IP multimedia subsystems

- **S-CSCF** (Serving Call State Control Function)

An S-CSCF provides session control services for a user. It maintains session states for a registered user's on-going sessions and performs the following main tasks.

- **Registration**: An S-CSCF can act as a SIP Registrar to accept users' SIP registration requests and make users' registration and location information available to location servers such as the HSS (Home Subscriber Server).
- **Session Control**: An S-CSCF can perform SIP session control functions for a registered user. Relay SIP requests and responses between calling and called parties.
- **Proxy Server**: An S-CSCF may act as a SIP Proxy Server that relays SIP messages between users and other CSCFs or SIP servers.
- **Interactions with Application Servers**: An S-CSCF acts as the interface to application servers and other IP or legacy service platforms.
- **Other functions**: An S-CSCF performs a range of other functions not mentioned above. For example, it provides service-related event notifications to users and generates Call Detail Records (CDRs) needed for accounting and billing

- **P-CSCF**

A P-CSCF is a mobile's first contact point inside a local (or visited) IMS. It acts as a SIP Proxy Server. In other words, the P-CSCF accepts SIP requests from the mobiles and then either serves these requests internally or forwards them to other servers. The P-CSCF includes a Policy Control Function (PCF) that controls the policy regarding how bearers in the packet-switched network should be used. The P-CSCF performs the following specific functions:

- Forward SIP REGISTER request from a mobile to the mobile's home network. If an I-CSCF is used in the mobile's home network, the P-CSCF will forward the SIP REGISTER request to the I-CSCF. Otherwise, the P-CSCF will forward the SIP REGISTER request to an S-CSCF in the mobile's home network. The P-CSCF determines where a SIP REGISTER request should be forwarded based on the home domain name in the SIP REGISTER Request received from the mobile.
- Forward other SIP messages from a mobile to a SIP server (e.g. the mobile's S-CSCF in the mobile's home network). The P-CSCF determines to which SIP server the messages should be forwarded based on the result of the SIP registration process.
- Forward SIP messages from the network to a mobile.
- Compression and decompression of SIP messages. Compression is required to minimize the air-interface time.
- Perform necessary modifications to the SIP requests before forwarding them to other network entities.
- Maintain a security association with the mobile.
- Detect emergency session.
- Create CDRs.

- **I-CSCF**

An I-CSCF is an optional function that can be used to hide an operator networks internal structure from an external network when an I-CSCF is used. It serves as a central contact point within an operator's network for all sessions destined to a subscriber of that network or a roaming user currently visiting that network. Its main function is to select an S-CSCF for a user's session, route SIP requests to the selected S-CSCF. The I-CSCF selects an S-CSCF based primarily on the following information:

- Capabilities required by the user.
- Capabilities and availability of the S-CSCF and
- Topological information, such as the location of an S-CSCF and the location of the users P-CSCFs if they are in the same operators network as the S-CSCF.

- **The Databases: (HSS and SLF)**

The Home Subscriber Server (HSS) is the central repository for user-related information. Technically, the HSS is an evolution of the HLR (Home Location Register), which is a GSM node. The HSS contains all the user-related subscription data required to handle multimedia sessions. These data include, among other items, location information, security information (including both authentication and authorization information), user profile information (including the services that the user is subscribed to), and the S-CSCF (Serving-CSCF) allocated to the user. A network may contain more than one HSS, in case the number of subscribers is too high to be handled by a single HSS. In any case, all the data related to a particular user are stored in a single HSS. Networks with a single HSS do not need an SLF. On the other hand, networks with more than one HSS do require an SLF.

The SLF is a simple database that maps users addresses to HSSs. A node that queries the SLF, with a user address as the input, obtains the HSS that contains all the information related to that user as the output. Both the HSS and the SLF implement the Diameter protocol (RFC 3588) with an IMS-specific Diameter application.

- **The Media Gateway Control Function (MGCF) and the IM Media Gateway (IM-MGW)** are responsible for signaling and media inter-working, respectively, between the PS domain and circuit-switched networks (e.g. PSTN).

- **The Multimedia Resource Function Processor (MRFP)** controls the bearer on the M_b interface including processing the media streams (e.g. audio transcoding). The Multimedia Resource Function Controller (MRFC) interprets signaling information from an S-CSCF or a SIP-based Application Server and controls the media streams resources in the MRFP accordingly.

The Breakout Gateway Control Function (BGCF) selects to which PSTN network a session should be forwarded. IT will then be responsible for forwarding the session signaling to the appropriate MGCF and BGCF in the destination PSTN network.

➢ **Reference Interfaces:**

The main interface in the IMS can be grouped into the following categories:-

Interface for SIP-based signaling and service control:

These include interfaces M_g, M_i, M_j, M_k, M_r, and M_w, which all use SIP as the signaling protocol.

- ✓ Interface $\mathbf{M_g}$ allows CSCF to interact with MGCF.
- ✓ Interface $\mathbf{M_i}$ allows a CSCF to forward session signaling to a BGCF so that the session can be forwarded to PSTN networks.
- ✓ Interface $\mathbf{M_j}$ allows a BGCF to forward a session signaling to a selected MGCF that will carry the session to the PSTN.
- ✓ Interface $\mathbf{M_k}$ allows a BGCF to forward session signaling to another BGCF.
- ✓ Interface $\mathbf{M_r}$ allows an S-CSCF to interact with an MRFC.
- ✓ Interface $\mathbf{M_w}$ allows an I-CSCF to direct mobile- terminated session to an S-CSCF.

Interface for controlling media gateways: These include interfaces M_c & M_p,

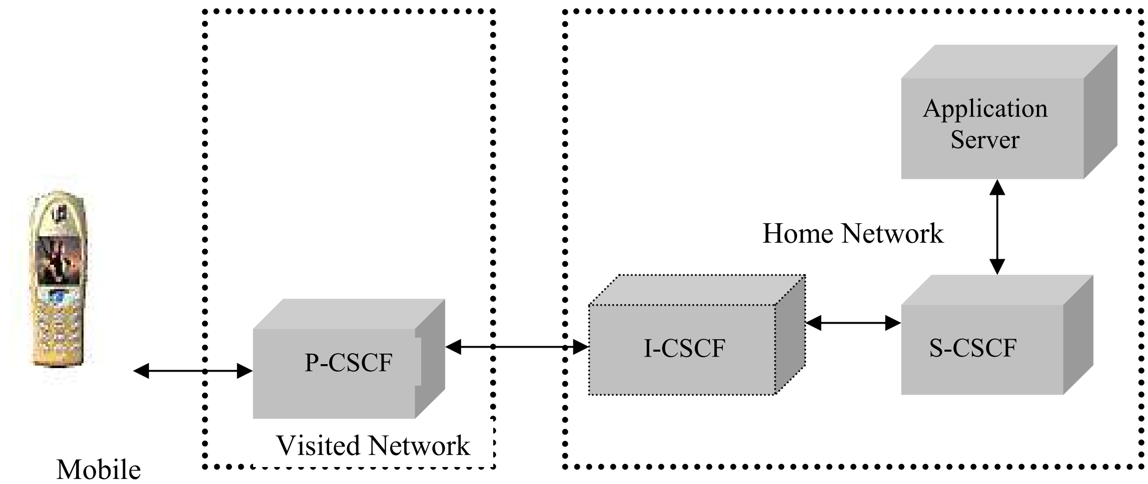
- ✓ Interface M_c allows a signaling gateway to control media gateway. For example, it is used between an MGCF and an IM-MGW, between an MSC Server and a CS-MGW, or between a GMSC Server and a CS-MGW.
- ✓ Interface M_p, allows an MRFC to control media stream resources provided by an MRFP. Signaling over interfaces M_c and M_p uses the H.248 /MegaCo Protocol.

Interfaces with the Information Servers: Interfaces C_x between the CSCF and the HSS allows the CSCF to retrieve from the HSS mobility and routing information regarding a mobile user so that the CSCF can determine how to process a user's sessions. Signaling over C_x interface uses the Diameter Protocol.

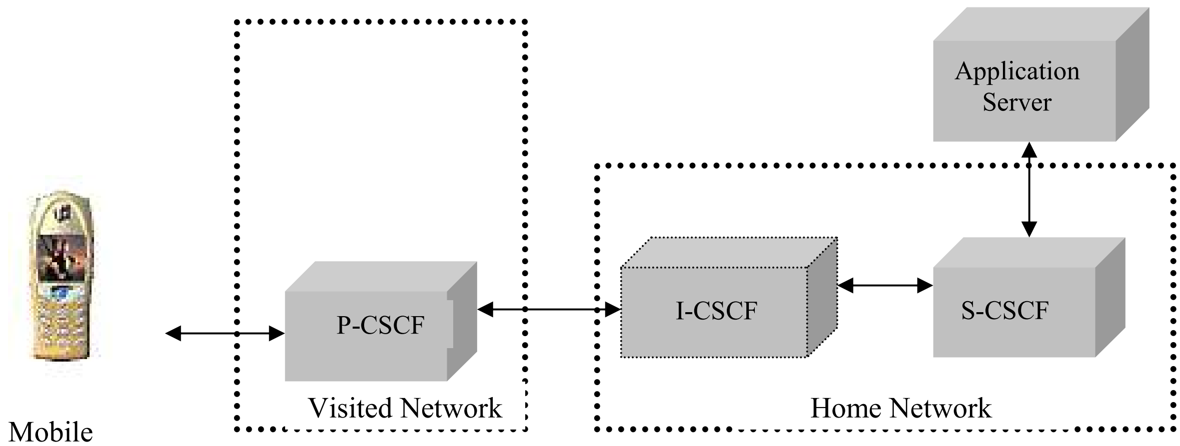
Interface with external networks: These include interfaces M_b, M_m, and C_o.

- ✓ Interface $\mathbf{M_b}$, is the standard IP routing and transport interface with external IP networks. The interface M_b may be identical to the G_i interface.
- ✓ Interface $\mathbf{M_m}$ is a standard IP-based signaling interface that handles signaling inter-working between the IMS and external IP networks.
- ✓ Interface $\mathbf{G_o}$ allows a PCF to apply policy control over the bearer usage in the PS domain.

Service Architecture:



(a) Service platform in mobile's home network



(b) External Service platform

Fig. 2: 3GPP service architecture for real-time services

With both service architectures, the initial SIP request from a mobile travels from the originating mobile to the visited P-CSCF first, which then forwards the request to the I-CSCF (if used) in the originating mobiles home network. This I-CSCF selects an S-CSCF in the home network for this user session and forwards the SIP request to session will travel directly between the visited P-CSCF and the S-CSCF in the mobiles home network.

The S-CSCF is responsible for interfacing with internal and external service platforms as illustrated in Fig. 3. There are three types of standardized platforms:

(1) SIP application server
(2) Open Service Access (OSA) Service Capability Server (SCS) and
(3) IP Multimedia Service Switching Function (IM-SSF).

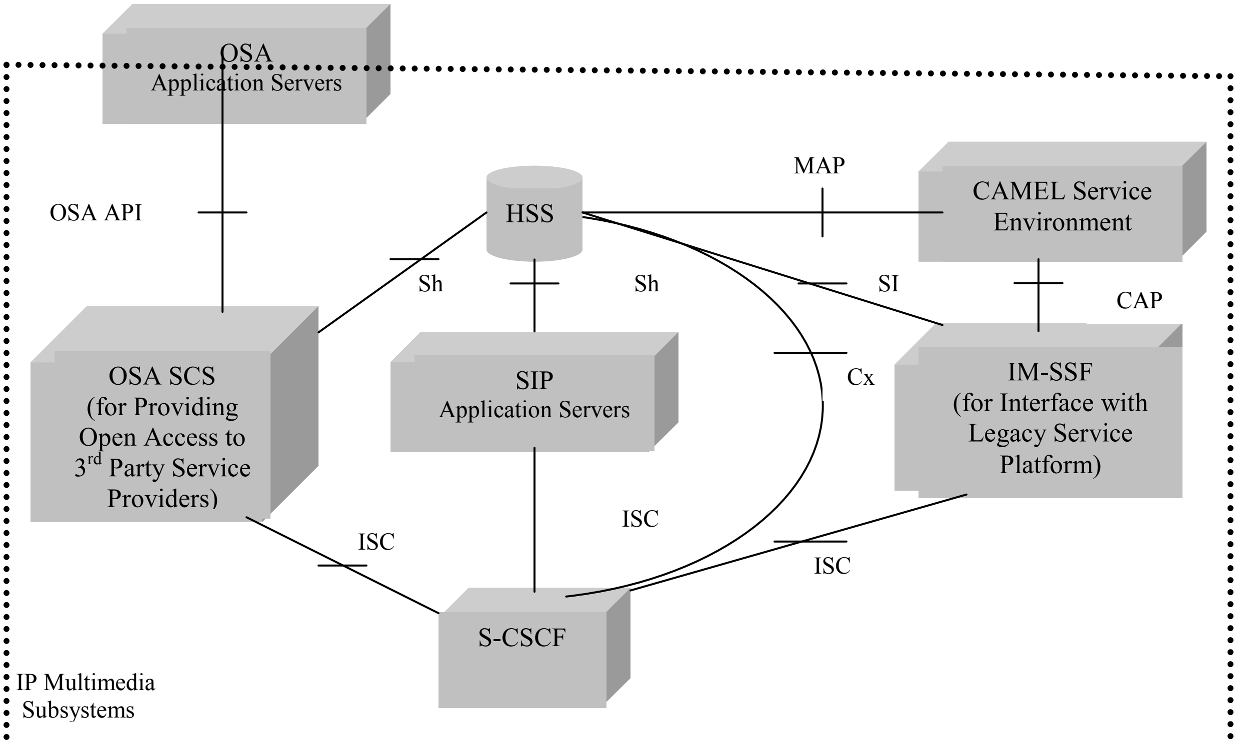


Fig. 3 Interactions between S-CSCF and service platforms

The services offered by them are value-added services (VAS or operator-specific services. The S-CSCF uses the same interface, IMS Service Control (ISC) interface, to interface with all service platforms. The signaling protocol over the ISC interface is SIP. The OSA SCS and IM-SSF by themselves are not application servers. Instead, they are gateways to other service environments. As depicted in Fig. 3, the OSA SCS and IM-SSF interface to the OSA application server and CAMEL Service Environment (CSE), respectively. From the perspective of the S-CSCF, however, they all exhibit the same ISC interface behavior. The services are briefly described:

- **SIP Application Server**:

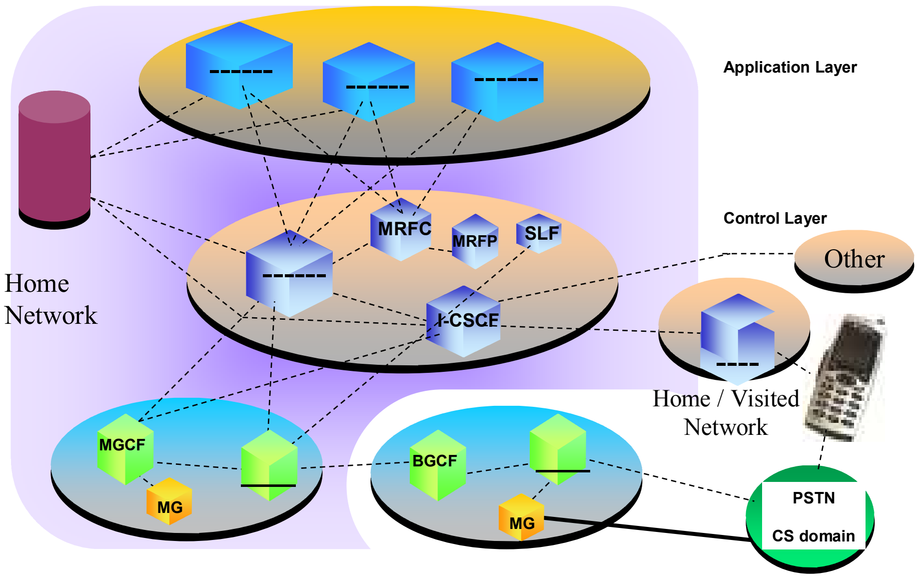
In addition to session control, a SIP server can also provide various value-added services. A lightweight SIP-based server enables the CSCF to utilize the SIP-based services and interact with the ISP application servers without additional components.

- **CAMEL Service Environment (CSE):**

The CSE provides legacy Intelligent Network (IN) services. It allows operators leverage existing infrastructure for IMS services. As specified earlier, the CSCF interacts with CSE through IM-SSF. The IM-SSF hosts the CAMEL features and interfaces with CSE by CAP (CAMEL Application Part).

- **OSA Application Server**:

Applications may be developed by a third party that is not the owner of the network infrastructure. The OSA application server framework provides a standardized way for a third party to secure access to the IMS. The OSA reference architecture defines an OSA Application Server as the service execution environment for third-party applications. The OSA application server then interfaces with the CSCF through the OSA SCS by OSA API (Application Programming Interface).



Practice Questions

Q1 In the above diagram note down the missing elements as per 3GPP IMS architecture?
Q2 The core of IMS is combining the best of two worlds ______ & ______ industry.
Q3 A ______ is a mobile's first contact point inside a local (or visited) IMS.
Q4 The IM-SSF hosts the ______ features and interfaces with CSE by CAP
Q5. The Multimedia Resource Function Controller (MRFC) interprets signaling information from an S-CSCF or a SIP-based Application Server and controls the media streams resources in the MRFP accordingly.(True/False)
Q6. The _____ contains all the user-related subscription data required to handle multimedia sessions.(HSS/SIP Application Server)
Q7.The statement that Networks with a single HSS do not need an SLF is false.(T/F)
Q8. An _______can act as a SIP Registrar to accept users' SIP registration requests and make users' registration and location information.(S-CSCF/P-CSCF/I-CSCF)
Q9When MGCF shall be required?
Q10.IMS stands for______________________ .

xxxx