TEC
TELECOMMUNICATION ENGINEERING CENTRE

NEWSLETTER
टी ई सी संचारिका

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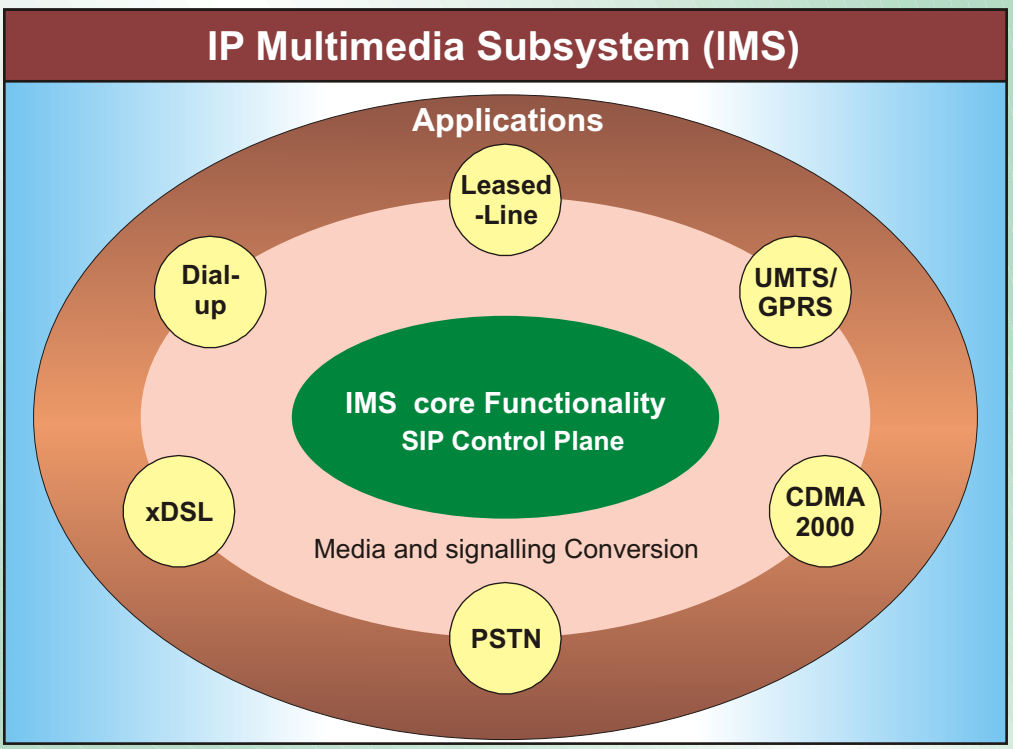

IN THIS ISSUE

Foreword

I have been assigned the responsibility of Sr. DDG, Telecom Engineering Centre (TEC) w.e.f. 17th March, 2006.

It would be my endeavour to make TEC more effective in making useful contribution to the development of telecom in India.

TEC has been attempting to create awareness about technological developments through TEC Newsletter. I take this opportunity to convey my best wishes to our readers. We invite your suggestions.

(R.N.Padukone)
Sr. DDG, TEC

IP Multimedia Subsystem (IMS)

IMS is standardised by 3GPP (3rd Generation Partnership Project) and subsequently adopted and endorsed by 3GPP2, for 3G mobile networks. International Standards Organisations like ITU, ETSI etc. are working to adopt it for other types of access.

IMS enables IP-based peer-to-peer connectivity and is envisaged to be the heart of Next generation Network (NGN). The IMS establishes packet data connections between end points in a carrier-class, chargeable way. Once established, the IP connection can be used to exchange all types of communication media, including voice, video, content and more. The IMS provides a full suite of network capabilities for authentication of clients, network-to-network interfaces and administrative functions such as charging.

Session Initiation Protocol (SIP)

Session Initiation Protocol (SIP) is a signalling protocol that handles the setup, modification and tear-down of multi-media sessions. SIP is defined by the IETF and has been widely used throughout the Internet. Like HTTP, SIP is a text-based client-to-server protocol. SIP has been adopted by the 3GPP as the session management protocol for IMS. The Building Blocks of an IP conversation are RTP (Real-time Transport Protocol), which provides transport of media streams and Session Initiation Protocol (SIP) which sets up those media streams.

In the simplest case, SIP session establishment can be a relatively straightforward process. First, the caller sends a SIP Invite message to the IP address of the called party. This invite message includes the proposed RTP technical parameters (e.g. Codec Type). The endpoints exchange additional SIP messages (including confirmation of negotiated RTP parameters) and the RTP session begins directly between the endpoints.

SIP requests and responses are sent to particular addresses known as SIP URLs (Uniform Resource Locators). These addresses take the form of user@host, similar to e-mail addresses. Since SIP deals with voice and can interwork with present telephone networks, SIP enables the user portion of the SIP address to be a telephone number, e.g. SIP :23329540@telco.net.

A typical IMS call flow

When a SIP client issues a find and connect request, the IMS finds the relevant terminals and connects them via IP.

At the heart of IMS is call/session control Function (CSCF), which is split into three parts- the proxy-CSCF (P-CSCF), interrogating-CSCF (I-CSCF) and serving-CSCF (S-CSCF). The CSCF components of the originating and terminating networks work together to resolve the called party address into an IP address.

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3G Network Architecture

The architecture of a 3GPP PLMN is shown in the Figure below. It consists of one or more Radio Access Network (RANs) interconnected via a Core Network (CN).

A RAN provides radio resources (e.g. radio channels, bandwidth) for users to access the CN.

A GSM/EDGE RAN (GERAN) is divided into Base Station Subsystems (BSS). Each BSS consists of one or multiple Base Transceiver Stations (BTSs) and Base Station Controllers (BSCs). A BTS handles the air interface. It handles signalling and speech processing over the air interface. A BSC controls the radio connections toward the mobile terminals as well as the wireline connections toward the CN. Each BSC can control one or more BTSs.

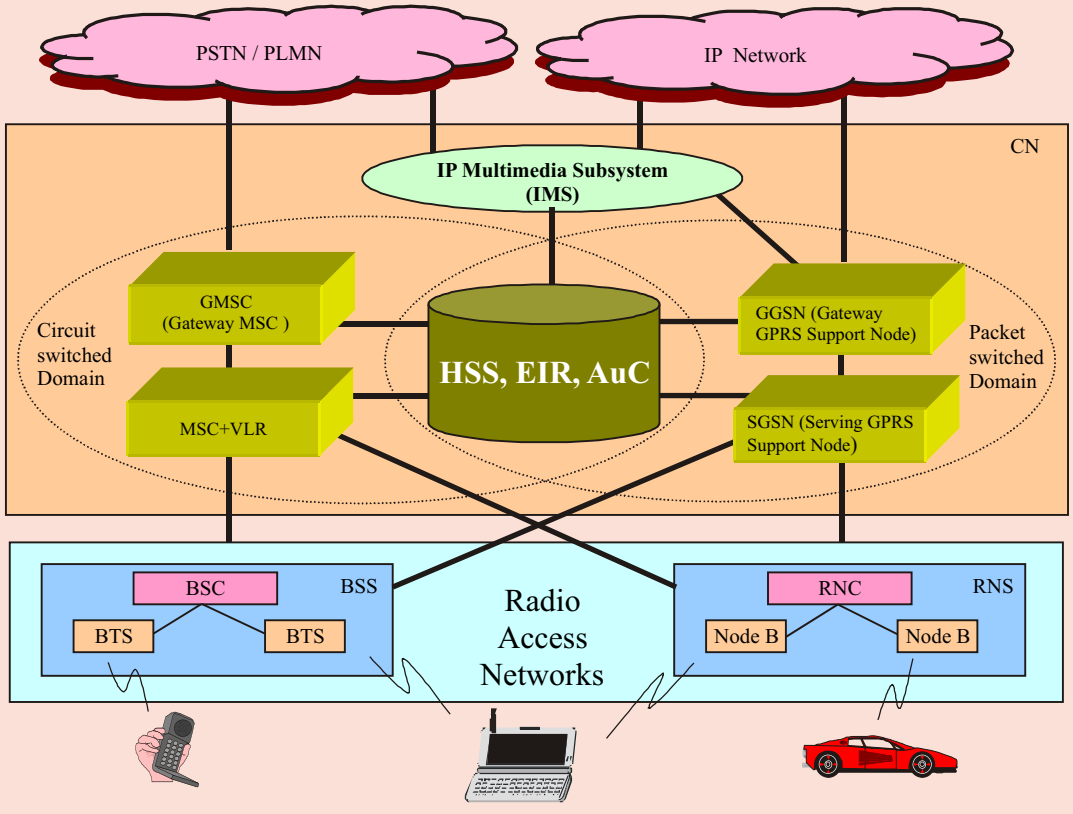
A UMTS Terrestrial Radio Access network (UTRAN) is divided into Radio Network Subsystems (RNS). Each RNS consists of one or multiple Node Bs controlled by a Radio Network Controller (RNC). A Node B is a wireless base station, which is analogous to a BTS in GSM and it provides the air interface with mobile terminals. An RNC, which is analogous to a BSC in GSM, controls the radios connections towards the mobile terminals and the wireline interfaces with the CN.

The CN implements the capabilities for supporting both circuit-switched and packet-switched communication services to mobile users. These communication services include basic services and advanced packet services.

The CN is divided into the following functional building blocks :

- Circuit-Switched (CS) Domain
- Packet-Switched (PS) Domain
- IP Multimedia Subsystem (IMS)
- Databases

Each RAN routes circuit-switched traffic to the CS CN domain and routes packet-switched traffic to the PS CN domain.



The P-CSCF is the SIP proxy which is the entry point for SIP terminals. The SIP terminals receive the address of the P-CSCF of the visited network at start time. Then, the terminals send their SIP messages to the P-CSCF. P-CSCF analyses the request. Extracting the terminal IMS address (and its domain) from the SIP message, it forwards the request to the home I-CSCF.

The I-CSCF is the SIP proxy at the entry point of the IMS domain. The I-CSCF receives the SIP messages from subscriber terminals through the P-CSCF of the domain or through the P-CSCF of other domains (in the case of roaming), e.g. a call by a subscriber. The I-CSCF also receives the SIP messages from the P-CSCF of other domains (e.g. reception of a call from outside). Then, the I-CSCF gets the user status from the HSS and forwards the SIP message to an S-CSCF. To illustrate the CSCF component roles, consider the establishment of a conversation (audio or video) from one mobile phone to another as shown in figure 1.

For ease of understanding, it is useful to think of call establishment in three phases:

Service Control provides address resolution so that the SIP Invite message gets to the right destination user terminal.

QoS Establishment authorises and reserves the necessary network resources.

Ring and Answer, i.e. completion of call, including commitment of resources.

Within the Service Control phase, the SIP Invite message, having traversed the originating networks (visited and home), comes to the I-CSCF and S-CSCF of terminating home network. The S-CSCF has to apply the Service Control logic for the termination of the call to the right end device. For that to happen, it needs to know about the destination terminal and its associated P-CSCF, which occurs by a process called registration.

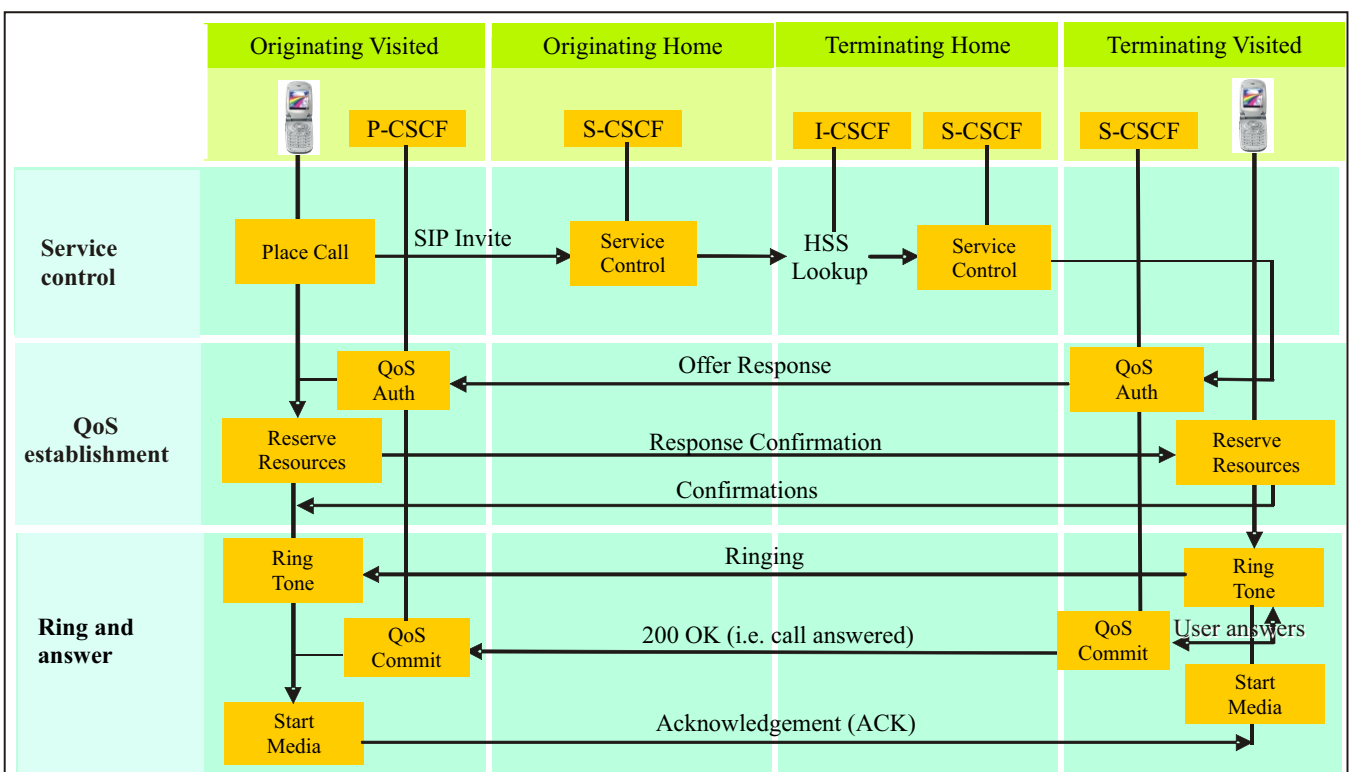


Fig. 1 - Example of IMS call flow

The registration process

The registration is a process by which a terminal in a visited network locates the S-CSCF it should register with. Such a process occurs when the terminal is switched on or enters network coverage.

First, the terminal must discover the address of a P-CSCF, which it typically does in conjunction with obtaining its IP address e.g. via DHCP (Dynamic Host Configuration protocol). The P-CSCF is always the first point of contact in the network for the IMS client. After registration, it will forward SIP Invites received from the client directly to the correct S-CSCF. However, during registration, it does not yet know which S-CSCF is the right one, so first it uses the domain name associated with mobile phone to look up an I-CSCF for that domain via DNS (Domain Name Server). The primary function of the I-CSCF is to refer a SIP message to the right S-CSCF based on user profile.

User profile information in IMS is always stored in a Home Subscriber Server (HSS). While in the current version of IMS there can only be one HSS per IMS user, users from different domains may be stored on different HSSs. In that case, the I-CSCF refers to a subscriber Location Function (SLF) to determine which HSS is associated with the domain of the user registering. It then looks up the right HSS to determine the right S-CSCF and forwards the registration message to the correct S-CSCF.

After completion of the registration process, both the S-CSCF and the P-CSCF maintain the association with that user so that later, when the terminal originates a call, the P-CSCF forwards the SIP Invite directly to the correct S-CSCF. In other words, for outgoing calls, the I-CSCF is not involved after the registration process.

The same is not true for incoming calls. Because it is the I-CSCF that is listed in DNS, it is to the I-CSCF that an originating S-CSCF will send SIP messages for a particular domain. As for the registration process, the I-CSCF looks up the HSS to find the right S-CSCF to handle termination of the call.

Important UMTS (Universal Mobile Telecommunications System) Releases

Release 97 (1997):	Known as 2.5G, this release introduced GPRS for data delivery over GSM (2G).
Release 99 (1999, UMTS R3 3GPP):	First release of the (3G) UMTS standard, in 1999. Included W-CDMA.
UMTS Release 4 (2001):	Separated the system into Circuit Switched and Packet Switched domains
UMTS Release 5 (March 2003):	First IMS Release, introduced the IP Multimedia Subsystem (IMS) as control structure of the Packet Domain, based on SIP for call control and mandatory IPv6.
UMTS Release 6 (December 2004 to March 2005):	Includes some leftover IMS issues from Release 5.0, such as QOS Improvements, plus enhancements to use SIP, and event-based charging.

As per one technical review (www.bcr.com), it now appears that 3GPP operator systems will use IMS for Web data and signalling, while maintaining circuit-switched voice bearers for the foreseeable future. IMS Rel-5 is likely to be limited to experimentation in laboratory settings, while production deployments of IMS, when they become feasible, will probably include devices with an aggregate of IMS functions selected by vendors from Rel-5, Rel-6 and Rel-7.

Clearly then, the complete vision of IMS invoked by network operators and equipment suppliers lies in the future, but exactly when is unknown. IMS is considered as complex and costly. In this vision, IMS is described as the means to integrate voice and data services over a packet-based infrastructure, to deliver end-to-end QOS for toll-quality VOIP, and to converge wireline and wireless infrastructures under a common set of end-to-end signalling and billing mechanisms.

Source : Internet, Technology white paper : "IMS Inside the Enterprise" by P.Carden, E.Darmois, P.Tournassoud

IMPORTANT ACTIVITIES OF TEC DURING DECEMBER 2005 TO MARCH 2006

Preparation of GRs/IRs

Following GRs/IRs and Technical documents were issued:

Multi Services Provisioning Platform (MSPP) with RPR Packet Ring capability for ADM & DXC
User Equipment for WCDMA (3G)
Voucher Management and Roaming Recharge Service for Mobile Networks
Integrated Access Device (IAD)
BTS Shelter
ISP(Internet Service Provider) Application
VoIP Protocol Analyser

Revised GRs/IRs

Synchronous Supply Unit (SSU)
Digital Subscriber Line Test Set (DSL)
Copper Pair Tester for DSL working
Wired Access Network
National Standard for V5.1 and V5.2 Interface
RF fixed Attenuator (DC-40 GHz)
Base Station Antennas in 824-894 MHz, 1880-1900 MHz and 1920-2170 MHz Frequency Bands

Tests and Field trials

Tests/field trials have been carried out for:

'OneIndia' tariff Plan of BSNL
IN system (Alcatel) of BSNL West Zone
32K SIM Card for BSNL of M/s Axalto and M/s Syscom Corporation Ltd.
Router of Cisco
15 GHz Microwave Radio of Nokia
Non-usability of RF Signal

Other Activities

Manufacturer Forum conducted for

- WEB Cache Engine
- Uninterrupted Power Supply (UPS) system
- VRLA batteries based on GEL Technology
- Standard on Electromagnetic Compatibility for Telecommunication equipment
- Fibre Distribution Management System
- Metal Free Optical Fibre Cable
- National SCCP standards and IN system & service description

Technical evaluation of Wi-Fi Hot Spot project
Technical guidelines for implementation of 'OneIndia tariff Plan' in BSNL

Approvals issued by TEC during the period December 2005 to March 2006

Interface Approvals.....	41
Service Test Certificates.....	119
Total	160

Approvals issued by TEC upto 31.03.2006

Type Approvals.....	7018
Interface Approvals.....	4262
Service Test Certificates.....	1877
Grand Total	13157

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