3GPP: IMS

(IP Multimedia Subsystem)

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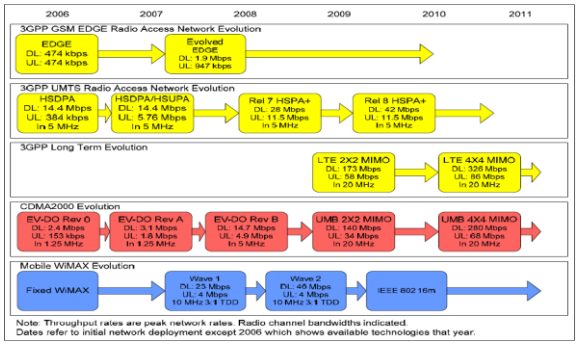
2012

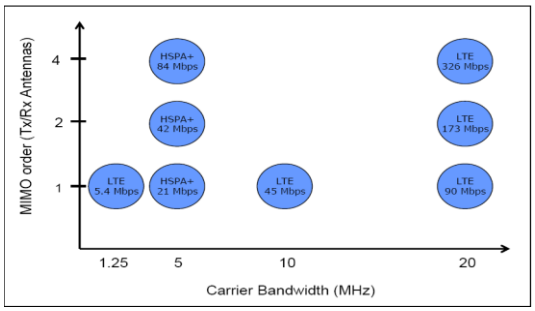
Outline

- 3GPP Evolution
- SIP Architecture
- Mobility Management
- SIP and 3G Networks

TDMA, CDMA, OFDMA

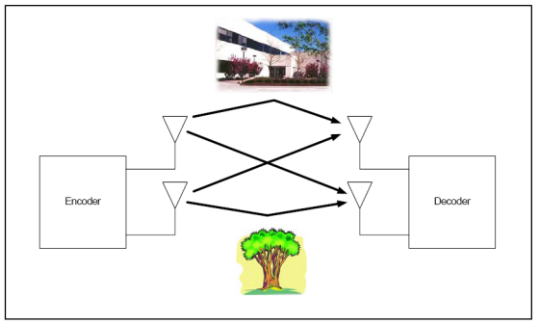


HSPA+, LTE Possible Peak Downlink Rate

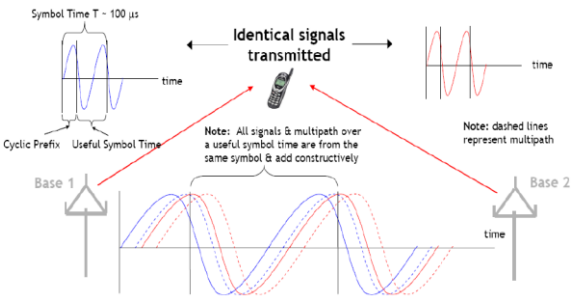


MIMO

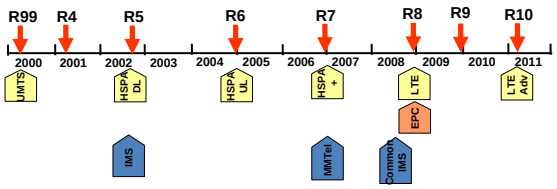
Figure 32: MIMO Using Multiple Paths to Boost Throughput and Capacity



Mobile Multicast/Broadcast TV

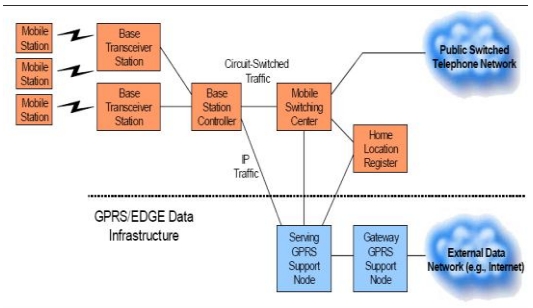


3GPP Release

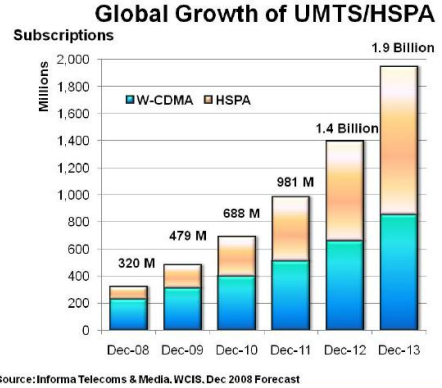


Releases cover the areas of:
. Accesses (GSM, EDGE, HSPA, UMTS, LTE, LTE-Advanced, etc.)
. Core Network (GSM Core, EPC)
. Services (IMS, MMTel)

GPRS/EDGE Data Infrastructure



Global Growth of UMTS/HSPA



Source: Informa Telecoms & Media, WCIS, Dec 2008 Forecast

UMTS Voice and Data Traffic

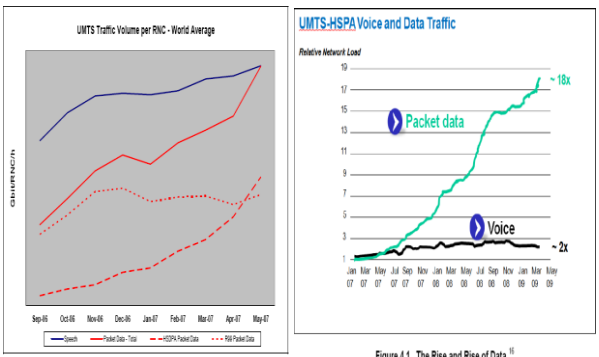
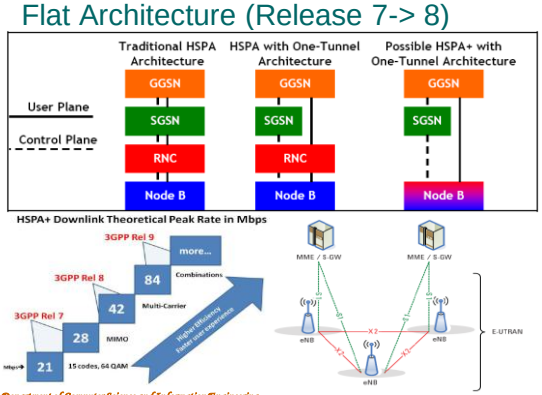
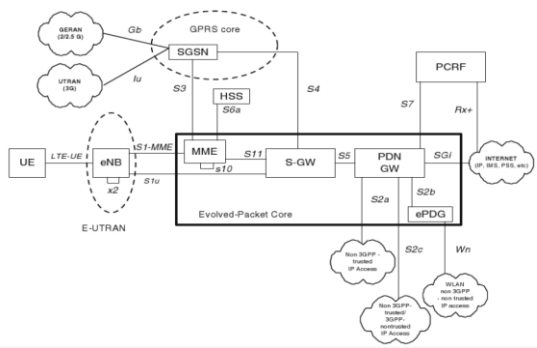


Figure 4.1. The Rise and Rise of Data.[1]

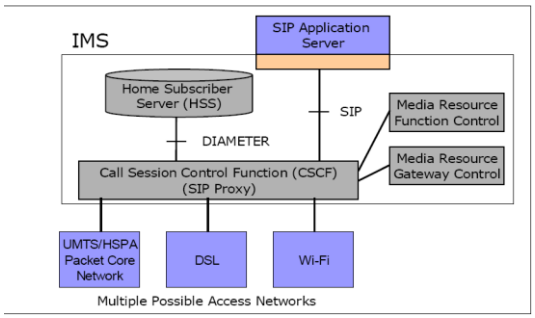
Flat Architecture (Release 7-> 8)



EPC (Evolved-Packet Core)



IMS-IP Multimedia Subsystem



IMS: IP Multimedia Subsystem

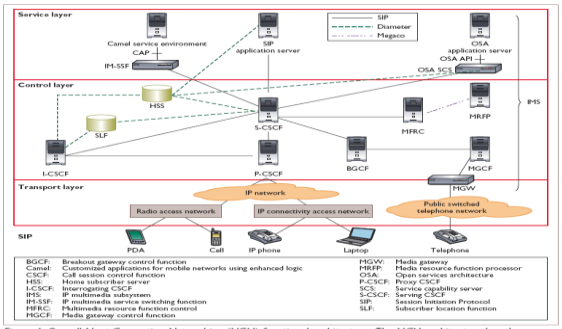


Figure 1. Overall Next-Generation Networking (NGN) functional architecture. The NGN architecture has three layers — transport, control, and service — with the IP multimedia subsystem (IMS) at the architecture's core.

MMTel (Multimedia Telephony)

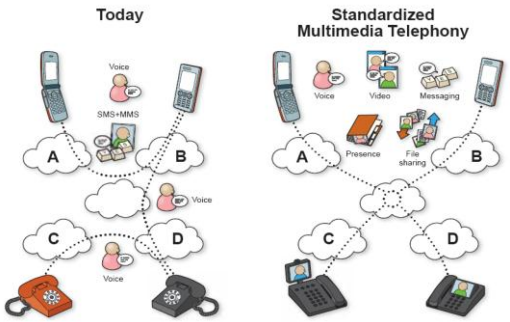


Figure 3: Looking forward: the MMTel standard enables operators to interconnect services

Session Initiation Protocol (SIP)

Session Initiation Protocol

- SIP is originally proposed by Columbia University and is specified by IETF.
- SIP is an end-to-end application-layer protocol
  - Establish, modify and terminate interactive multimedia sessions, e.g., VoIP and video conference, between SIP-based users.
  - Signaling protocol.
  - Client-Server framework.
- H.323 is a alternative signaling protocol to support VoIP.

Microsoft Voice .NET Services

Features of SIP

- Text-based
  - Easy implementation in Java or Perl
    - JSIP open source library
  - Easy debugging
  - Flexible and extensible
- Less signaling comparing to H.323
  - QoS
- Transport-layer independence
  - UDP is commonly used.
- Forking a call request
  - Call forwarding
  - Parallel rings at different places

H.323

The H.323 standard

The first version of H.323, which was intended for multimedia communications over local-area networks (LANs), appeared in 1996. Many found it to be lacking the functions needed for supporting VoIP in a broader environment. Consequently it was revised and H.323 version 2'—'Packet-based multimedia communications systems'—was released in 1998. This version of H.323 has received more support than its predecessor, particularly among those network operators and equipment vendors who have a background in more traditional telephony. H.323 is not an individual protocol, rather it is a complete, vertically integrated suite of protocols that defines every component of a VoIP network—terminals, gateways, gatekeepers, MCUs (Multipoint Control Units) and servers with other features. Amongst others, H.323 uses the following standards:

- Q.931 for call set-up
- H.225 for call signalling
- H.245 for exchanging information on terminal capabilities and creation of media channels
- H.245 for RAS-registration, admission and status (RAS) control
- RTP/RTCP for sequencing audio and video packets
- G.711/712, a codec specification
- T.120 for data conferencing.

All these protocols—involving dozens of back-and-forth messages—are called upon in setting up a simple point-to-point voice call. In contrast, SIP is a simple protocol that specifies only what it needs to. For example, SIP works with RTP but does not mandate it.

- Introduction to SIP
- SIP Architecture
- Mobility management
- SIP and 3G Networks

Four SIP Logical Entities

- User agent
- Proxy Server
- Registrar
- Redirect Server

User Agent

- User applications
- Both software and hardware

Type of SIP Servers

- Proxy Server
  - Application layer router used to relay SIP messages.
- Registrar
  - Accept registration request from user agent.
- Redirect Server
  - Redirects caller to other servers.

Typically, "SIP server" implements the functionality of Proxy, Registrar and Redirect Servers.

SIP Addressing

- SIP give you a globally reachable address.
  - Email-like address.
    - sip: leonard@a.ntu.edu.tw
    - sip: 82828888@a.ntu.edu.tw
- User agents bind this address to Registrar by using SIP REGISTER message.
- Each user agent communicates with one another by using this address.

SIP messages

Generic-message = start-line
*message-header
CRLF
[message-body]
start-line = Request-Line | Status-Line

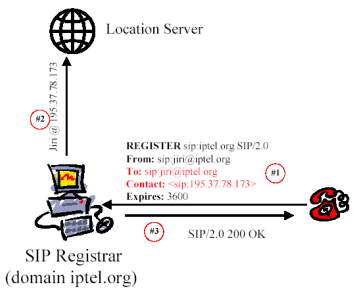
Request Line →

SIP message	Description
INVITE	Invites a user to a call
ACK	Used to facilitate reliable message exchange for INVITEs
OPTIONS	Solicits information about a server's capabilities
BYE	Terminates a connection between users or declines a call
CANCEL	Terminates a request, or search, for a user
REGISTER	Registers a user's current location
INFO	Used for mid-session signalling

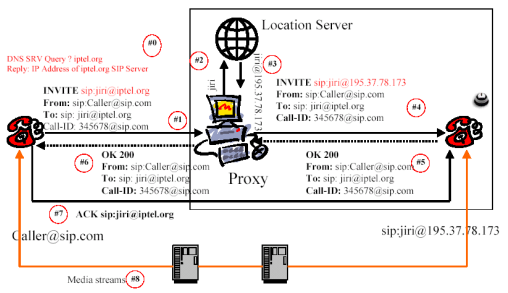
Class	Description	Example
1xx	Informational: request received, continuing to process the request	100 Trying, 180 Ringing
2xx	Successful: the action was successfully received, understood and accepted	200 OK
3xx	Redirection: further action needs to be taken in order to complete the request	302 Moved Temporarily
4xx	Client Error: the request contains bad syntax or cannot be fulfilled at this server	404 Not Found
5xx	Server Error: the server failed to fulfil an apparently valid request	501 Not Implemented
6xx	Global Failure: the request cannot be fulfilled at any server	603 Decline

↑ Status Line

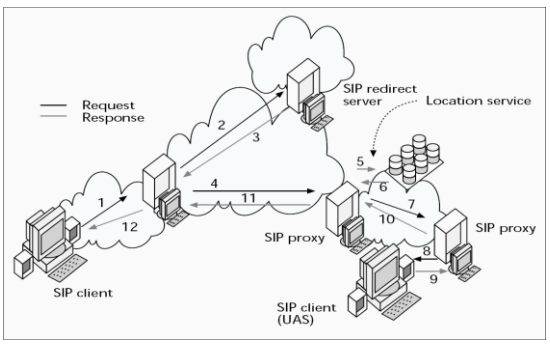
Example: SIP Registration



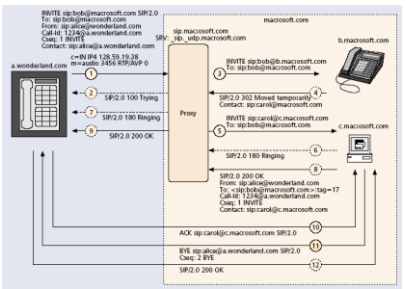
Example: Session Establishment



Routing Information



Example: Session Forwarding



Session Description Protocol (SDP)

- The message body of SIP
- SDP is used to describe a multimedia session

```
u = http://www.ietf.org
e = g.bell@bell-telephone.com
c = IN IP4 132.151.1.19
m = audio 3456 RTP/AVP 96
a = rtpmap:96 VDVI/8000/1
m = video 3458 RTP/AVP 31
m = application 32416 udp wb
a = orient:portrait
```


RTP, RTCP, and RTSP

- Real Time Transport Protocol (RTP)
  - Encode and decode media stream
  - Recover the possible loss and jitter
- Real Time Control Protocol (RTCP)
  - QoS feedback
  - …
- Real Time Streaming Protocol (RTSP)
  - Control stored media
  - VCR remote control
  - Support play, record , pause, fast forward, and etc.

RTSP protocol session

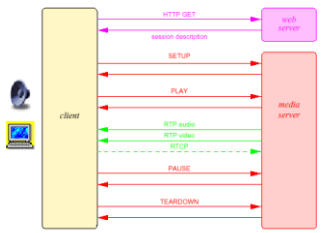
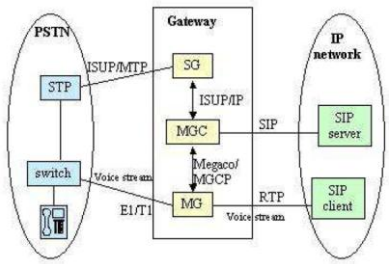


Figure 6: RTSP protocol session

SIP Interworking with the SS7



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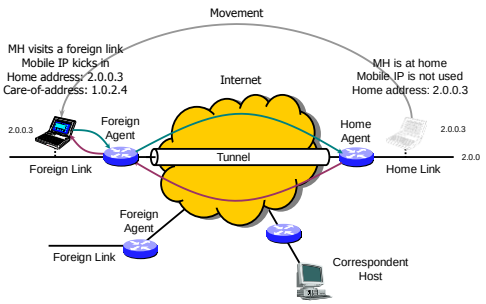
Wireless Technologies Convergence

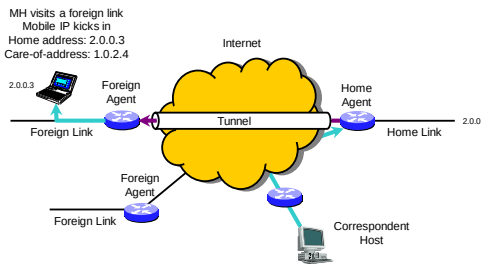
Mobility Management

- Mobility Classification
  - Roaming
  - Macro-mobility
    - Domain mobility
  - Micro-mobility
    - Subnet mobility
- Solutions
  - Network layer solution: Mobile IP
  - Application layer solution: SIP

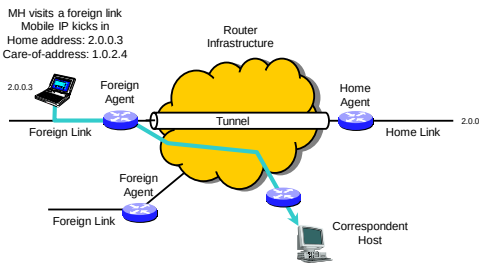
Mobile IPv4: Registration Example



Mobile IPv4: CH-to-MH Routing Example



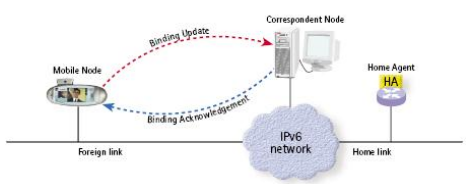
Mobile IPv4: MH-to-CH Routing Example



Mobile IPv4

- Triangle route problem
- Micro-mobility improvement
  - Cellular IP, Campbell in Column University.
  - Regional Registration, Perkins, Nokia Center.
  - …

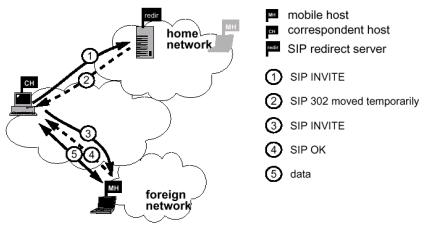
Mobile IPv6: Binding Update



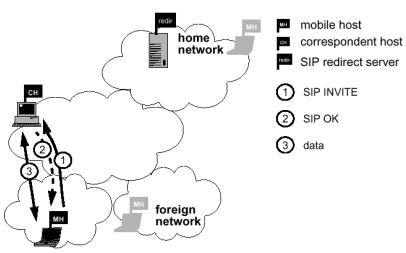
Application Layer Mobility Using SIP

- Terminal Mobility
- Session Mobility

Terminal Mobility

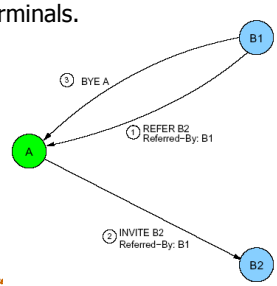


Terminal Mobility

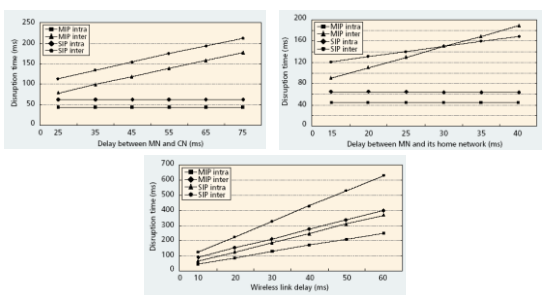


Session Mobility

- Allow a user to maintain a media session even while changing terminals.



Comparison



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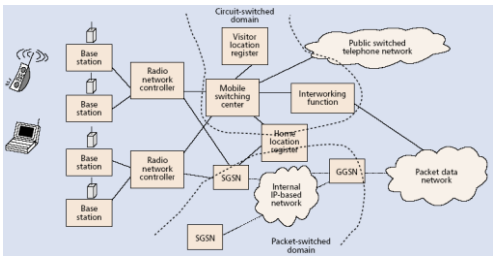
3G UMTS



Pure IP connectivity vs. Dedicated Multimedia subsystem

- Some mechanisms should be defined in 3G to support multimedia session transfers?
- Market Perspective
  - Subscriber perspective
  - Network operator perspective
  - Third-party service provider perspective

Subscriber Perspective

- Advantages
  - It is free and flexible to choose applications.
  - Reuse application in wired-networks
- Disadvantage
  - Trouble to choose the application and service provider.
  - The demand of service package and one bill.
  - Some application may lose QoS guarantee.

Network Operator Perspective

- Advantages
  - Operators may not have experience in IP multimedia applications. They only focus in the IP connectivity.
- Disadvantages
  - Circuit-switch revenue will be decayed.
  - Loss possible revenue for paving basic IP multimedia application, e.g., VoIP.
  - Issue of customer dissatisfaction for IP multimedia applications.

Third-Party Service Provider Perspective

- Advantages
  - They don't have to bother the peculiarity of wireless networks. They don not need extensive knowledge of wireless telecommunication networks and protocols.
- Disadvantages
  - They are unable to take advantage of the wireless network, e.g., user location information.

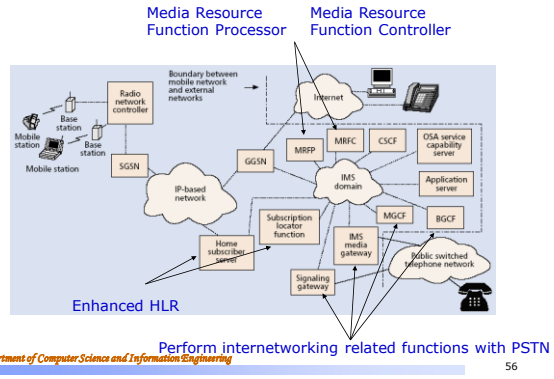
IP Multimedia Subsystem (IMS)

- Appear in Release 5 and beyond
- IMS comprises the network elements for control of multimedia sessions.
- Network operator provides both
  - IP connectivity
  - Multimedia session management

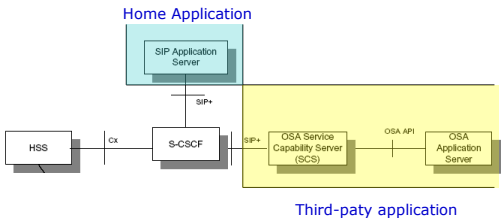
Basic Add-in Features

- Call State Control Function (CSCF)
  - Provisioning of call control for IP multimedia applications. P-CSCF, I-CSCF, S-CSCF.
- Open Service Access (OSA)
  - Third-party are expected to stimulate innovative application, taking advantage of knowing the capabilities provided by wireless network providers.

IP Multimedia Subsystem



Serving-CSCF



S-CSCF

- Session control.
- Retrieve the information from HSS.
- Connect to Application Servers.
- Each user agent needs to attach a S-CSCF before setup a session.
- Analog to Registrar in SIP.

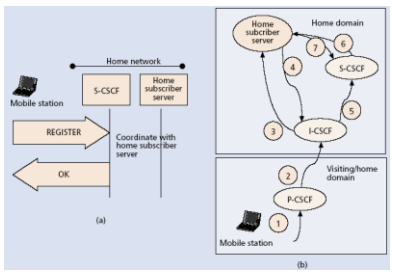
Proxy-CSCF (P-CSCF)

- The first contact point within the IMS.
- Mobile node communicate with S-CSCF via P-CSCF. Direct communication with S-CSCF is not allowed.
  - Integrity protection of SIP signaling.
  - Compression due to sparse wireless resource (Sigcom).
  - Inspect SIP signaling if the mobile node is in a visited network.

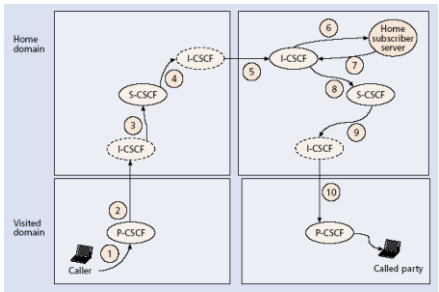
Interrogating-CSCF (I-CSCF)

- Entry Point in a network operator.
- Hide the configurations, topology and capacity from outside.
- Analog to Proxy and redirect servers in SIP.

Registration



Session Setup



MMTel and Circuit Switch internetworking

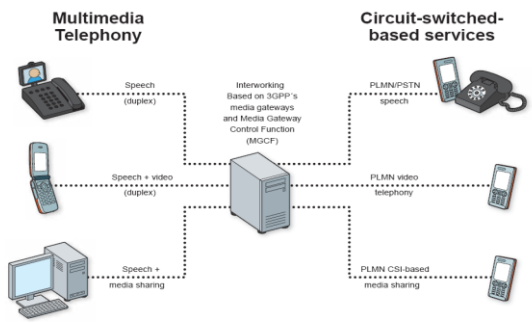


Figure 4: Important interworking scenarios between MMTel and standard circuit-switched services

Reference

Mobile IP: Charles E. Perkins http://people.nokia.net/~charliep
SIP: http://www.cs.columbia.edu/sip
IMS: 3GPP TS 23.228 v2.0 http://www.3gpp.org/ftp