IP-based Multimedia Subsystem (IMS)

Motivation
NGN/IMS Architecture
Role of CSCF
SIP-based Call Flows
UMTS Network Architecture, Rel. 5

Goals:

- combine the growth of the Internet with the growth in mobile communications
- convergence of voice, video, messaging, data and web-based technologies
- conform to IETF “Internet standards” as much as possible
- enable PLMN operators to offer multimedia services based on Internet applications, services and protocols
- service development by PLMN operators and other third party suppliers including those in the Internet space using the mechanisms provided by the Internet and the IMS

Concept:

- IMS comprises all CN elements providing multimedia services (signaling and bearer related network element)
- IP multimedia services are based on IETF-defined session control capability
- IP multimedia services use the PS domain
- IMS provides a set of PS services equivalent to the relevant subset of CS services

Source: TS 23.228
Changing Telecoms Trends

- **Fixed line** usage is reducing dramatically for “classical” services.
- **Mobile use** is increasing steadily even though penetration is already high.
- **Broadband Internet** deployment shows a rapid growth trend.
Application and Services Market Trends

- **Media Convergence** - *Multiple Play*
  - Dual Play: High-Speed Internet & Fixed Line
  - Triple Play: Dual Play + TV
  - Quadruple Play: Triple Play + Wireless

- **Fixed Mobile Convergence**
  - Dual Mode connectivity
    - Cellular / Cordless (DECT, ADSL/Bluetooth)
    - WLAN / WWAN

- **MVNO - Mobile Virtual Network Operator**
  - Wireless Service Reseller, wholesales access from wireless operators
  - Discount & Lifestyle MVNO’s
  - Segment, Product, Utilization Driven

- **M-Commerce - Electronic Commerce using Mobile Phones**
  - Leverage ubiquity of mobile phones to make transactions
  - Current payment methods: premium calling #'s, phone bill invoice, credit card
  - Strong interest in key industries: banking, sports & entertainment, travel, retail

- **Multimedia - use of several media types to convey information**
  - Effective information delivery across many disciplines: art, education, telecommunications, medicine
  - IMS enables multimedia services for mobile users
    - VoIP

- **Presence - Always on, always connected**
  - Combine Mobility & Reachability
  - Effectively bring Popularity of IM to mobile phones (AOL, Yahoo!, MSN, Skype)
  - Opportunity for standardization & interworking based on SIP/SIMPLE

With seamless blending across wireline and wireless, a service provider can gain larger share of the “telecom wallet”
How Can We Build “Service Enabled” Networks to Support Sophisticated Service Interworking?

Service Architecture Requirements
- Common User Interface for all services for an end-customer
- Open application server interfaces to allow a rich set of applications
- Separate subscriber data for a consistent, maintainable subscriber data
- Common session control to support service interworking
- Support for “mobility”
- Support of the full set of access networks and endpoints (POTS, VoIP, …)
IMS – A Standard Service Architecture

- VoIP Telephony and Multimedia Services Architecture
- Defined with Open Standard Interfaces -> 3GPP and 3GPP2
- Based on IETF Protocols (SIP, RTP, ..)
- Designed for Both Wireless and Wireline Networks
- A Solution for Service Transparency
- Capable of Interworking with PSTN and Legacy IN Based Services

CSCF – Call Session Control Function
HSS – Home Subscriber Server
What is IMS?

- The “Internet Protocol Multimedia Subsystem” (IMS) has been defined for 3GPP networks
  - IMS standards define a network domain dedicated to the control and integration of multimedia services.
  - IMS is defined by 3GPP from Release 5 onwards (since 2001)
  - Considered to be the standard for fixed and mobile Internet-based telephony by Operators.
- IMS builds on IETF protocols
  - Based upon SIP, SDP, RTP, COPS and Diameter protocols
  - 3GPP have enhanced these IETF protocols for mobility

- **IMS in short**
  - *Open-systems architecture that supports a range of IP-based services over the PS domain, employing both wireless and fixed access technologies*
What does IMS provide?

- **Services and Control**
  - Adds call session control to the packet network (GPRS)
  - Enables peer-to-peer real-time services - such as voice, video - over a packet-switched domain

- **Mixed Multimedia**
  - Ability to pick and mix various multimedia flows in single or multiple sessions
  - Can handle real-time voice, video, data

- **Connectivity Independence**
  - Provides access to IP based services independent of the connectivity network: mobile (3GPP’s UMTS, 3GPP2’s CDMA2000) and fixed networks (TISPAN’s NGN)
IMS is Access Independent (in Theory)

- **3GPP Defined Radio Accesses For Packet**
  - LTE
  - HSPA+
  - HSPA
  - TD-SCDMA
  - WCDMA
  - EDGE
  - GPRS

- **Non-3GPP Defined Radio Accesses**
  - 802.11
  - 802.x
  - 3GPP2 MDN

- **Fixed Accesses**
  - DSL
  - Cable
Organizations using IMS

- Cellular Access to IMS
- Wireline Access to IMS
- WLAN Access to IMS

From left to right:
- CSI
- VCC
- Presence
- GLMS
- PoC
- Messaging
- Multimedia Telephony
- Multimedia Telephony
- Multimedia Telephony

UMTS Networks Andreas Mitschele-Thiel, Michael Söllner WS 2007
Reference Architecture for IMS

Source: 3GPP 23.821-1.0.1 (document is outdated)

Signalling and Data Transfer Interface

*) those elements are duplicated for figure layout purpose only, they belong to the same logical element in the reference model
NGN Architecture Based on 3GPP/3GPP2 IMS

AS - Application Servers – provides feature logic
- Session Initiation Protocol (SIP) AS
- Open Service Access (OSA)
- Applications include: Telephony, IP Centrex, Push-to-talk, Instant messaging.
NGN Architecture Based on 3GPP/3GPP2 IMS

**HSS - Home Subscriber Server**
- Centralized DB for user profiles
- DHCP, DNS, ENUM functions
- HLR successor
- Authentication Center for Security

**Network Resources (HSS)**
- Subscriber, GUP, Charging, DHCP, DNS, ENUM

**Support Systems**
- Billing Mediation, Fault Correlation, Operations, Maintenance, …
NGN Architecture Based on 3GPP/3GPP2 IMS

**Application Layer**
- Parlay Application
- Web Services
- SIP Application
- Parlay Gateway (OSA SCS)
- Service Broker (SCIM)

**Session Control Layer**
- S-CSCF
- I-CSCF
- P-CSCF
- BGCF

**Media and End Point Layer**
- Media Server
- MRFC
- SIP
- PDF
- IP Signalling Converter
- Session Border Controller
- Access Network
- SIP Endpoints
- Legac

**CSCF - Call Session Control Function**
- P-CSCF – Proxy: Entry point to IMS for devices
- I-CSCF – Interrogating: Entry point to IMS from other networks
- S-CSCF – Serving: Session control entity for endpoint devices

**BGCF - Breakout Gateway Control Function**
- Selects network to use for PSTN interworking

**PDF - Policy Decision Function**
- Authorizes QoS requests

**MRFC - Multi-Media Resource Function Control**
- Manages Resource Servers
NGN Architecture Based on 3GPP/3GPP2 IMS

- **MGCF - Media Gateway Control Function**
  - Controls MGW
- **MGW - Media Gateway**
  - Inter-works RTP/IP and PCM bearers
- **SG – Signalling Gateway**
  - Interworks SIP with SS7/PRI
“Databases” contain the following information:

- **Subscriber Profile**
  - Contains subscriber specific information that is used for service and feature authorization.

- **Dynamic Subscriber Information**
  - Current session registration data (i.e.: S-CSCF Address, access network)

- **Network Policy Rules**
  - For subscription resource usage, QoS, valid times and routes, geographical service area definitions, policy rules for the applications serving a user, etc.

- **Equipment Identity Register (EIR)**
  - Information such as records of stolen equipment.

- **IP address management (DHCP, DNS, ENUM)**

- **AAA functions (Authentication Authority, …)**
Call Session Control Function (CSCF)

- SIP Proxies used to manage SIP sessions
  - Coordinates with other network elements
  - Session control, feature control, resource allocation, …

- Three flavors of CSCFs
  - Serving CSCF (S-CSCF) - Session control entity for endpoint devices
  - Interrogating CSCF (I-CSCF) - Entry point to IMS from other networks
  - Proxy CSCF (P-CSCF) - Entry point to IMS for devices

- Functionally CSCFs follow Internet paradigms
  - P-CSCF → I-CSCF → S-CSCF
  - Stateless entities at network edge, stateful entities in core
  - Simple processing at edge, complex processing in core
  - Security and authentication requirements increase towards core
CSCF Roles – IMS Call

Called UE
Home IMS Network

I-CSCF* (THIG)
S-CSCF
I-CSCF* (THIG)
I-CSCF* (THIG)

Calling UE
Home IMS Network

I-CSCF* (THIG)
S-CSCF
I-CSCF* (THIG)
I-CSCF* (THIG)

Called UE
Visited Network

I-CSCF* (optional)
P-CSCF
Access Network
I-CSCF* (optional)
P-CSCF
Access Network

Calling UE
Visited Network

I-CSCF* (optional)
P-CSCF
Access Network
I-CSCF* (optional)
P-CSCF
Access Network

UE – User Equipment

* Note – Session Border Controllers (SBC)s will often be deployed with I/P-CSCFs at carrier boundaries

THIG: Optional Topology
Hiding Inter-network Gateway

UMTS Networks
Andreas Mitschele-Thiel, Michael Söllner
WS 2007
Proxy CSCF (P-CSCF)

- First contact point within the IMS for the subscriber
  - Well known address(es) within network
  - P-CSCF discovery can either be static or via DHCP
- Authentication and Authorization
  - Routes incoming requests based on registration status
    - Sends the SIP REGISTER request received from the UE to an I-CSCF determined using the home domain name
    - Sends SIP messages received from the UE to the SIP server S-CSCF, whose name the P-CSCF has received from registration
  - Rejects non-authorized requests
  - Authorize the bearer resources for the appropriate QoS level
    - PDF functionality integrated in release 5, separate entity in release 6
- Acts as a stateful SIP proxy
  - Generates CDR events
  - Can act as User Agent and terminate calls in abnormal situations
  - Detects and handles emergency session establishments
- SIP compression and decompression
  - For wireless access networks
Interrogating CSCF (I-CSCF)

- Initial contact point for incoming network connections
  - Well known address(es) within network
  - Query HSS for the address of S-CSCF to handle call
  - Selects S-CSCF for a user performing SIP registration
    - Provides S-CSCF fan-out to support scalability
    - Selection can be static or dynamic
      (Based on current conditions and user location)
  - Routes request to proper S-CSCF or external network
- Acts as a stateless SIP proxy
  - Generates CDR events
- Provides Topology Hiding Inter-network Gateway (THIG)
  - Not required but provides valuable capabilities
  - Hides configuration, capacity, and topology of network from outside
Serving CSCF (S-CSCF)

- Registrar and Notification Server
  - Acts like an IETF RFC 3261 compliant Registrar
  - IETF RFC 3265 compliant event notifications, e.g., registration
  - Generally 1-1 binding between registered endpoint and S-CSCF

- Locally Stores Subscriber Data
  - The Serving CSCF retrieves the subscriber data from the HSS
  - Includes filter criteria information,
    - Which Application Servers to contact for specified events

- Session Control and Routing
  - Provides session control for the registered endpoint's sessions
  - Behaves as both SIP Proxy and User Agent
  - Generates session level CDRs

- Bearer Authorization
  - Ensures that media types and quantities indicated by SDP for a session are within boundaries of subscriber's profile

- Application Interaction
  - Interacts with Application Services platforms for the support of services
IMS/SIP Call Flow Examples
• **VoIP Application Requirements (three applications flows)**
  – SIP/SDP over UDP/IP
  – Voice payload over RTP/UDP/IP and
  – RTCP over UDP/IP
SIP FLOW

User Agent (Client) Sends SIP Requests

User Agent (Server) Receives SIP Requests

Proxy Server Determines Where to Send the Signaling Messages

Media Stream (RTP)

Signaling

Signaling
Typical SIP Architecture

SIP Client

SIP Proxy

SIP Redirect Server

Location Service

SIP Proxy

SIP Proxy

SIP Client (User Agent Server)

Request

Response
Methods (SIP Requests)

- **REGISTER**
  - Informs a SIP server about the location of a user

- **INVITE**
  - Invites a participant to a session/dialog

- **ACK**
  - For call acceptance

- **UPDATE**
  - Change media attributes of a session

- **OPTIONS**
  - Queries a participant about their capabilities

- **BYE**
  - Ends a client’s participation in a session/dialog

- **CANCEL**
  - Terminates a request

- **PRACK**
  - Confirm reliable delivery of an intermediate response

There are many more extensions!!
(e.g. SUBSCRIBE, NOTIFY, MESSAGE and more every day!)
Responses (SIP replies)

Divided into 6 classes:

1-xx: Informational
100 Trying
180 Ringing
...

2-xx: Successful
200 OK

3-xx: Redirection
300 Multiple Choices
301 Moved Temporarily
...

4-xx: Request Failure
400 Bad Request
482 Loop Detected
...

5-xx: Server Failure
500 Server Internal Error
501 Not Implemented
...

6-xx: Global Failure
603: Decline
606: Not Acceptable
...

All 2xx, 3xx, 4xx, and 5xx responses are **FINAL** (terminates the SIP transaction).

A 1xx is a **PROVISIONAL** SIP response.
A Basic Dialog...

Is it really this easy ... of course not!
A More Typical Dialog…!

INVITE [SDP offer]
100 Trying
407
ACK
INVITE [SDP offer]
100 Trying
180 Ringing [SDP answer]
PRACK [SDP offer]
200 OK (PRACK) [SDP answer]
200 OK (INVITE)
ACK
INVITE [SDP offer]
100 Trying
180 Ringing [SDP answer]
PRACK [SDP offer]
200 OK (PRACK) [SDP answer]
200 OK (INVITE)
ACK
SIP: Message Syntax

- Many header fields from http
- New ones (e.g. Via) are SIP specific
- Supports several payload types
  - SDP - Session
    Description Protocol: contains a media description
  - ISUP - encapsulated ISUP signaling message to bridge circuit networks

INVITE sip:dmoreland@lucent.com SIP/2.0
From: “Jim Calme” <sip:calme@lucent.com>
Subject: SIP Tutorial
To: “Doug Moreland” dmoreland@lucent.com
Via: SIP/2.0/UDP 128.3.4.5; branch=z9hG4bk40ac
Call-ID: 1997234505@il0015.calme.ih.lucent.com
CSeq: 4711
Max-Forward: 70
Content-type: application/sdp
Content-Length: 187

v=0
o=userid 53655765 2353687637 IN IP4 128.3.4.5
s=-
c=IN IP4 224.2.0.1
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
So how do we set up a call?

- In order to establish a call between two users, it is necessary to exchange media information. This information is provided by Session Description Protocol (SDP) carried in SIP messages.

- The SDP provides the following information:
  - The packet addresses and ports to be used.
  - The types of resources for the session (e.g., audio codecs)
  - Transport type (e.g., RTP/AVP)

- Bi-directional (send and receive) media paths are assumed unless otherwise indicated.
How is SDP exchanged?

- The originator’s SDP is “offered” in the INVITE and the terminator’s SDP is “answered” in a reliable response.
  - Since an unreliable transport (UDP) is typically used, extra measures must be used to ensure delivery.
  - Either a 200 OK or a “reliable” provisional response work.
- Session media may also be negotiated by offering several choices
- IMS allows two-round negotiation where the initial answer only returns the supported choices that were offered.
  - The originator then makes the final choice and offers that back to the terminator.
  - The terminator must then answer to confirm and provide its bearer details (e.g., address and port).
SIP Registration / Re-Registration

1. Initiate SIP Registration
2. Query DNS to obtain routing information for I-CSCF
3. Forward SIP REGISTER to Home Network
4. Retrieve information needed for S-CSCF Selection
5. Forward SIP REGISTER to S-CSCF
6. Retrieve and select Authentication Vector
7. Reject with Authentication Data
8. Re-initiate SIP Registration (steps 1 – 5)
9. Store S-CSCF Name
10. Retrieve Subscriber Profile and Filter Criteria
11. Register with AS(s) based on Filter Criteria
12. AS(s) retrieve Subscriber profile (if needed)
13. P-CSCF SUBSCRIBE, for de-registration
14. UE SUBSCRIBE, for de-registration
1. Initiate SIP Invitation
2. Retrieve Subscriber Profile (if needed)
3. Apply Service Logic
4. Retrieve Address of CLD Party Home Network and Forward INVITE.
5. Identify Registrar of CLD Party and Forward INVITE.
6. Retrieve Subscriber Profile (if needed)
7. Apply Service Logic
8. Forward INVITE to CLD Party
9. SDP Negotiation / Resource Reservation Control
10. Ringing / Alerting
11. Answer / Connect
IMS Subscriber to IMS Subscriber (Multiple Networks)

1. Initiate SIP Invitation
2. Retrieve Subscriber Profile (if needed)
3. Apply Service Logic
4. Retrieve Address of CLD Party Home Network and Forward INVITE.
5. Identify Registrar of CLD Party and Forward INVITE.
6. Retrieve Subscriber Profile (if needed)
7. Apply Service Logic
8. Forward INVITE to CLD Party
9. SDP Negotiation / Resource Reservation Control
10. Ringing / Alerting
11. Answer / Connect
IMS Subscriber to PSTN (Single BGCF)

1. Initiate SIP Invitation
2. Retrieve Subscriber Profile (if needed)
3. Apply Service Logic
4. Select network to access PSTN, and select MGCF
5. Seize trunk / determine media capabilities of MGW
6. SDP Negotiation / Resource Reservation Control
7. ISUP IAM
8. Ringing / Alerting
9. Answer / Connect

Calling Party Home Network

P-CSCF → S-CSCF → BGCF → MGCF → PSTN

H.248

SIP

RTP Stream

PCM

Call Control

Bearer
PSTN to IMS Subscriber

1. Incoming Call (ISUP IAM)
2. Seize Trunk and IP Port
3. Initiate SIP Invitation
4. Determine where the Subscriber is Registered
5. Forward SIP INVITE to S-CSCF
6. Retrieve Subscriber Profile (optional)
7. Service Logic (if needed)
8. Forward SIP INVITE to Called Party UE
9. SDP Negotiation / Resource Reservation Control
10. Alerting / Ringing
11. Connect / Answer
Simultaneous Ring and Presence Example

1. A calls B
2. Service Broker (SB) intercepts call initiation message*
3. SB determines that called party has TAS service
4. SB sends message on to TAS
5. TAS determines that called party has Simultaneous Ring (SR) feature
6. SB intercepts the SR call messages
7. SB sends query to Presence Server on called party
8. P determines status of called party endpoints
9. P returns results to SB
10. SB deletes call messages for cell phones with NOT ON state
11. Call is completed to B’s remaining endpoints

* Note: these “call messages” are typically SIP INVITES.
IMS Definitions and SIP

- 3GPP defines how IMS uses SIP in the following documents:
  - TS 24.229 – Base IMS call control
    - Lists which RFCs apply to IMS
    - Specifies IMS extensions to SIP headers and fields
    - Specifies IMS use of SIP in detail (information from RFCs not repeated)
    - What optional portions are mandatory with IMS
    - What additional rules may apply
  - TS 29.163 – Inter-working with circuit switched networks
  - TS 23.141 & TS 24.141 – Presence service
  - TS 24.147 – Conferencing service
  - TS 24.247 – Messaging service

- SIP Interfaces:
  - Gm Interface (UE – P-CSCF)
  - Mw Interface (CSCF – CSCF)
  - Mi Interface (CSCF – BGCF)
  - Mk Interface (BGCF – BGCF)
  - Mj Interface (BGCF – MGCF)
  - Mg Interface (CSCF – MGCF)
  - Mr Interface (CSCF – MRFC)
  - Mm Interface (CSCF – external SIP entity)
  - ISC Interface (S-CSCF – AS)
RFCs used by IMS for SIP

- RFC 3261 - (base) SIP protocol
- RFC 2976 - SIP INFO method
- RFC 3262 - Reliability of provisional responses in SIP
- RFC 3265 - SIP specific event notification
- RFC 3311 - SIP UPDATE method
- RFC 3312 - Integration of resource management and SIP
- RFC 3313 - Private SIP extensions for media authorization
- RFC 3323 - Privacy mechanism for SIP
- RFC 3325 - Private SIP extensions for network asserted identity
- RFC 3326 - Reason Header for SIP
- RFC 3327 - SIP extension header field for registering contacts
- RFC 3428 - SIP extension for instant messaging
- RFC 3455 - Private header extensions to SIP for 3GPP
- RFC 3515 - SIP REFER method
- RFC tbd - SIP extension header field for service route discovery
- RFC tbd - SIP event package for registrations
- RFC 2131 - Dynamic host configuration protocol
- RFC 2401 - Security architecture for the internet protocol
- RFC 2406 - IP Encapsulating Security for Payload (ESP)
- RFC 2617 - HTTP authentication
- RFC 2806 - URLs for telephone calls
- RFC 2916 - E.164 number and DNS
- RFC 3310 - HTTP digest authentication using AKA
- RFC 3315 - Dynamic host configuration protocol for IPv6
- RFC 3319 - DHCPv6 options for SIP servers
- RFC 3320 - Signaling Compression (SigComp) [3GPP]
- RFC 3329 - Security mechanism agreement for SIP [3GPP]
- RFC 3361 - DHCP for IPv4 option for SIP servers
- RFC 3420 - Internet Media Type message/sipfrag
- RFC 3485 - SIP and SDP static dictionary for SigComp
- RFC 3486 - Compressing SIP
Multimedia Broadcast/Multicast System (MBMS)
Motivation

- For some applications multiple users can receive the same multimedia data in the same time: It would beneficial for the network to transmit the data only once over a particular link.

- Cell Broadcast Services (CBS) can be used to transmit low bitrate data to particular cells (service areas).

- IP multicast as such cannot provide means for multiple subscribers to share radio or core network resources in UMTS system.
Multimedia Broadcast/Multicast System (MBMS)

Background

- R6 WI: Introduction of MBMS to UTRAN, TR 25.346
- Aims to offer an efficient way to transmit data from single source to multiple destinations over radio network → Network resources are saved
- MBMS is realised by addition of existing and new functional entities of the 3GPP architecture:
  - Broadcast Multicast Service Center connected to GGSN
- UTRAN issues for MBMS
  - Introduction of a point to multipoint bearer (ptm), only ptp available in R5
  - Counting procedures
  - New logical channels MTCH, MCCH/MSCH
  - No power control, service availability
  - Radio Resource Management (RRM)
  - Frequency layer convergence/dispersion (FLC/D)
  - Priority handling of services
Adding MBMS to a UMTS Network

Impact of adding MBMS

External Content Provider
Multicast/Broadcast Source

ATM/IP

nodeB

Uu

UE

nodeB

UE

internal Content Provider
Multicast/Broadcast Source

Broadcast Multicast
Service Center

Firewall

PSTN

External Content Provider
Multicast/Broadcast Source

packet
data

circuits
voice

Gi

Gmb

Gi

Gn

G GSN

SGSN

Iu-CS

RNC

Iu-PS

MSC

PSTN

IP

Internal Content Provider
Multicast/Broadcast Source

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Technical specifications for MBMS

- TS 22.146 “Requirements for MBMS”
- TS 23.246 “Architecture and functional description”
- TS 25.446 “UTRAN Interface Aspects”
- TS 26.346 “Codecs and Protocols”
- TS 29.846 “Procedure Description”
MBMS Service Modes

MBMS functions in two modes:

- **Broadcast Mode**
  - available without subscription for all UEs in a cell

- **Multicast Mode**
  - requires subscription

The multicast mode

- Subscription
- Service announcement
- Joining
- Session start
- MBMS notification
- Data transfer
- Session Stop
- Leaving

The broadcast mode

- Service announcement
- Session Start
- MBMS notification
- Data transfer
- Session Stop
MBMS issues

- Counting
  - Task: Determine number of subscribers in a cell to decide ptp/ptm
  - Problem: Number could be large
    - Note: UEs in all states could be subscribers (CELL_XXX are known)
  - Solution: Provide probability factors to control number of RACH responses

- New logical channels MTCH, MCCH/MSCH
  - Mapped on FACH mapped on S-CCPCH
  - BCCH carries MCCH info
  - MCCH carries MTCH bearer information
  - MSCH will contain schedule
MBMS issues

- No power control, no feedback

- Combat packet loss by
  - Higher layer FEC
  - Session Repetition (Recounting)
  - PTP repair service

- Radio Resource Management
  - Preferred Layer was introduced
  - Interlayer service management
  - UE diversion after session stop
MBMS issues

Service priority handling

- MBMS or DCH service could be prioritised
- Interrelated with UE capabilities
  (how many services can a UE receive at the time?)
- Priority handling is left to user decision
Summary

- IMS and MBMS support multimedia services
  - IMS for conversational services
    - VoIP
  - MBMS for Multicast/Broadcast
    - Ticker, Video Clip, Push-to-Talk (PTT)