IMS&SIP Introduction

Version: 01

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CONTENT

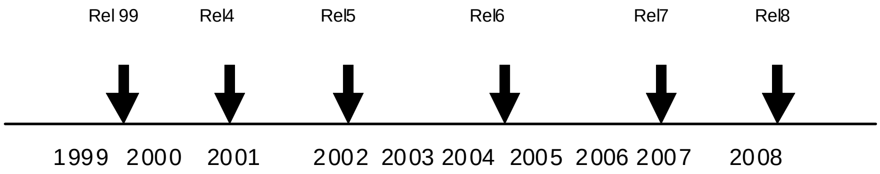
1 Introduction

This document is referring to the tendency of mobile telecommunication.
It is in the same time an introduction of IMS&SIP

2 Scope

In this document I am trying to specify the domain of interest.

3 UMTS Releases Overview



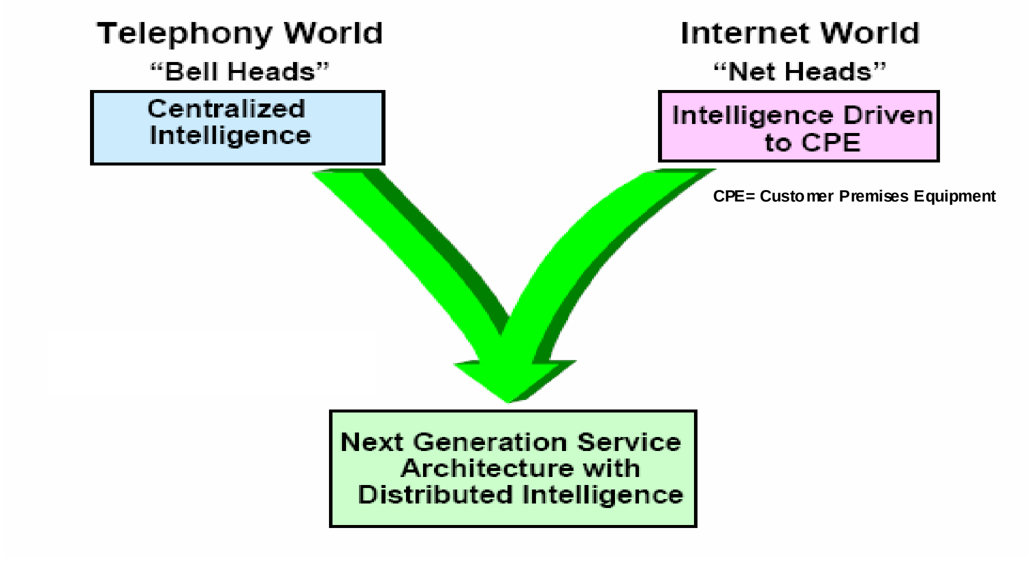
• Each year (almost) , a new Release of the UMTS standard is published
• Start from GSM
CS domain-GSM RAN
• initially < 10kb/s, evolved to today (EDGE) 384 kb/s
• GPRS
adds PS Domain, in parallel to CS Domain
Initially higher transmission rates than GSM (max 115 kb/s)
• can also be used with EDGE
Shared radio channel (DSCH)
=> more efficient usage of radio resources,
because bandwidth demands of e.g. web traffic are highly fluctuating
(user needs time to read page) and bursty
allows a direct connection to e.g. the Internet
charging per data volume possible
• in GSM always charging per time unit

4 UMTS Releases

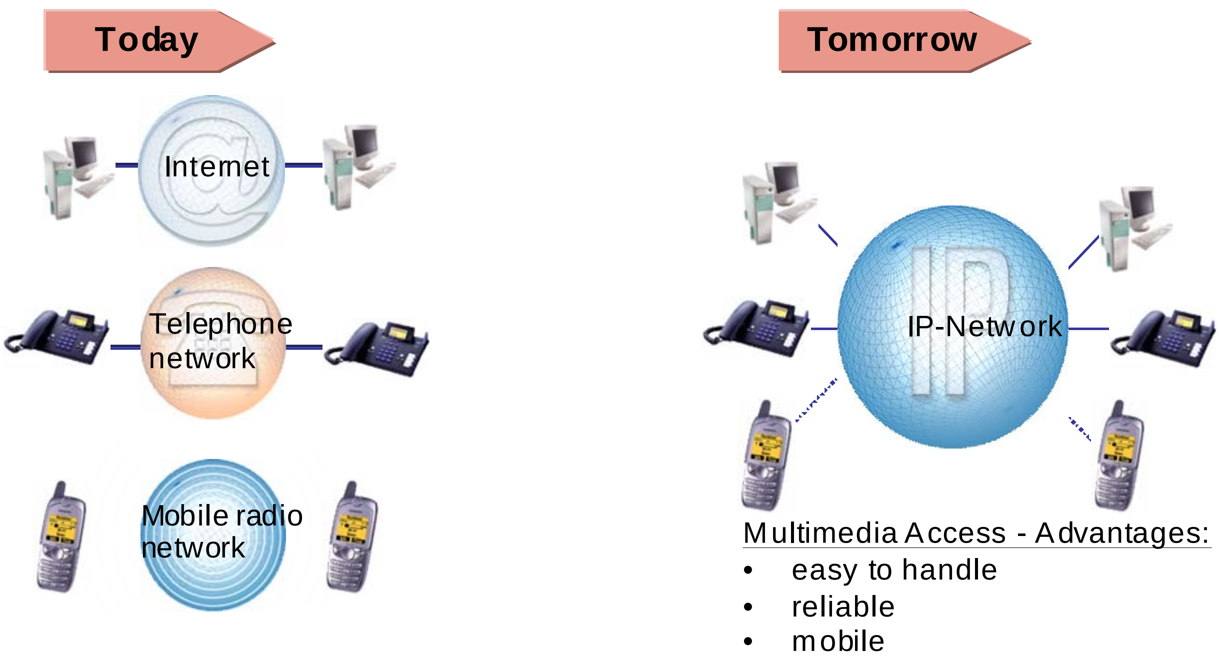
Below specified the main contribution of these releases:

UMTS Release 99 –>Frozen 12.1999
UTRAN &R99 Core
“Highspeed GSM” based on WCDMA and ATM(FDD and TDD)
Adopt all useful GSM R-99 services
UMTS Release 4 ->Frozen 03.2001
Bearer Independent Cs-Core; GERAN A/Gb mode (EGPRS)
“Cs Core Transport over IP”
Split CP from UP MGW and MSC-Server
UMTS Release 5 ->Frozen 06.2002
IP-UTRAN; IU-Flex; HSDPA;IMS
“All over IP”
RNC inter-working with Multiple Core Controllers (Multi Operators in R6)
High-speed Downlink Packet Access –up to 14 Mbit/s
IP Multi-Media Subsystem
UMTS Release 6–>Frozen Q2 -2005
HSUPA; MBMS
“Complementing IMS &3.5G”
HIGH-Speed Uplink Packet Access –up to 5.6Mbit/s
IP based Mobile Broadcast Multicast Services
UMTS Release 7 ->Frozen 06-2007
“3.75G”
HSPA+ Improvements; GERAN Performance (EGPRS2)
Performance improvements to HSPDA/HSUPA, Femtocell
Performance & QoS to support IMS,PS-Handover
UMTS Release 8 ->Expected Q4-2008
“LTE”
LTE/SAE- UTRAN Long Term Evolution and System Architecture Evolution
UMTS Release 9
“LTE Advances”

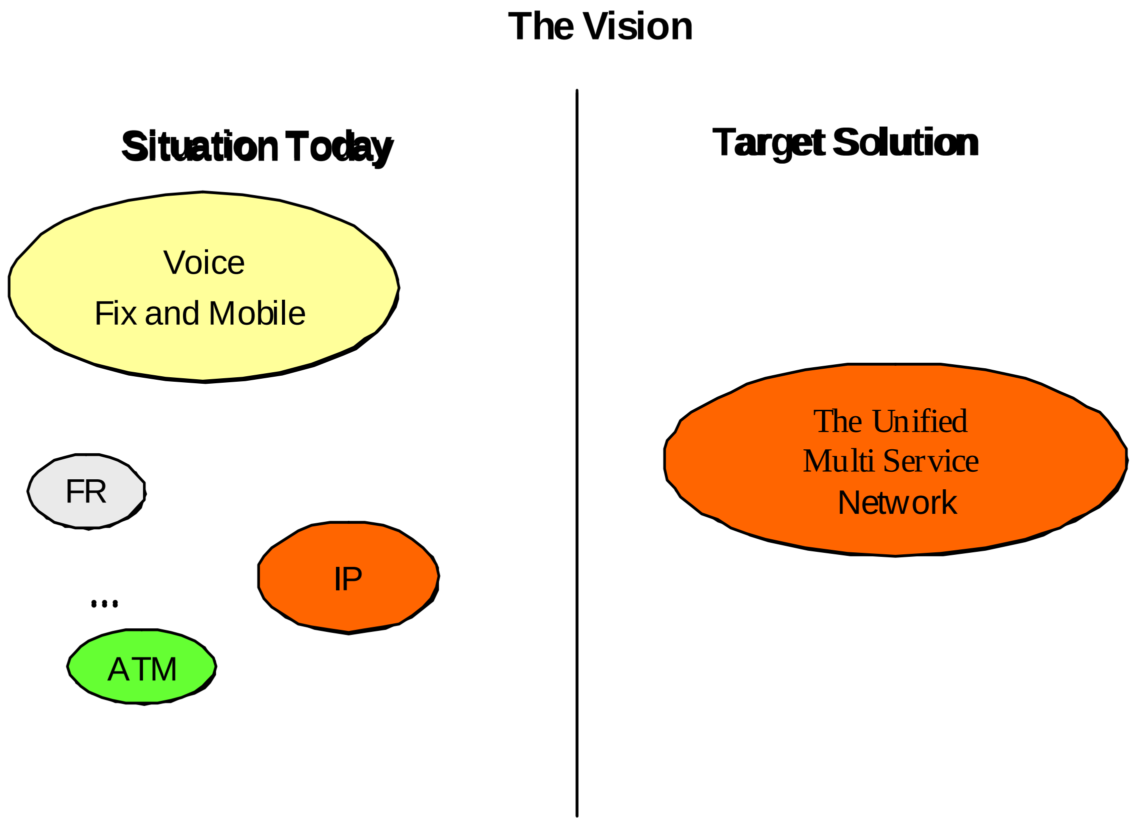
5 Convergence of Telephony World and Internet World



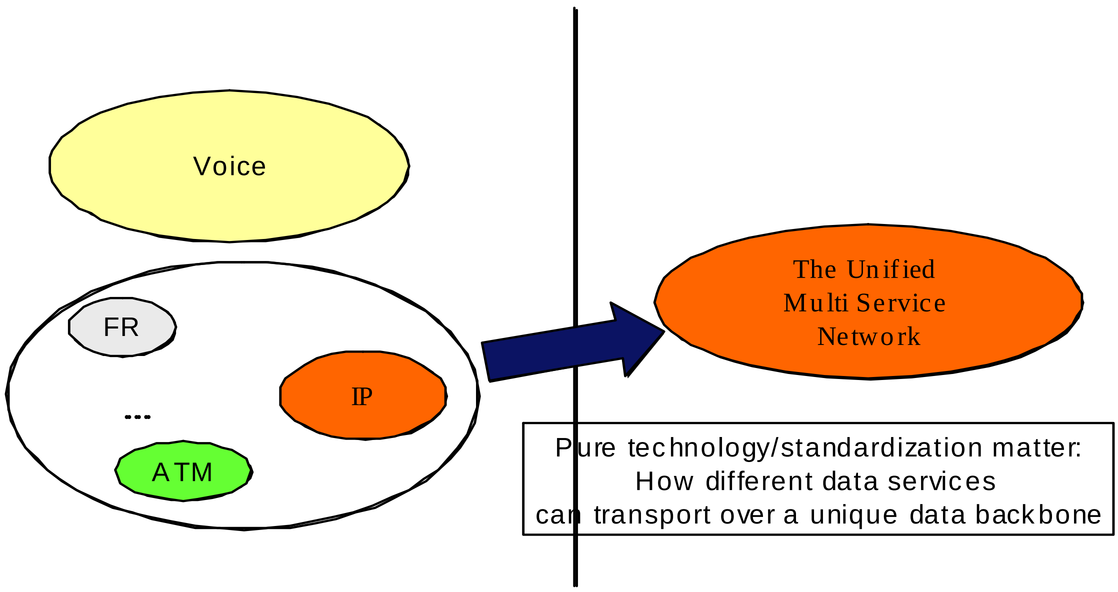
New wave **All over IP**



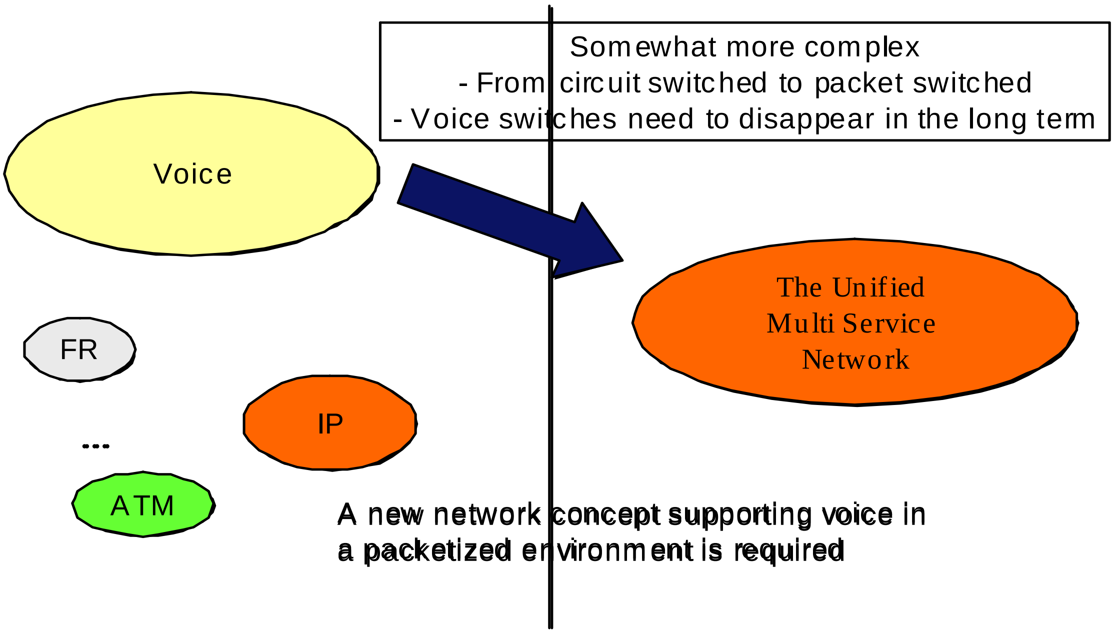
6 The Unified Network



The Data Migration



The Voice Migration



7 Key drivers of new development

Short Term objective:
Create new revenue possibilities

Removal of boundaries between voice and data opens the way to new kind of services
Can be realized relatively quickly with limited investments

Long Term objective:
Realize cost savings

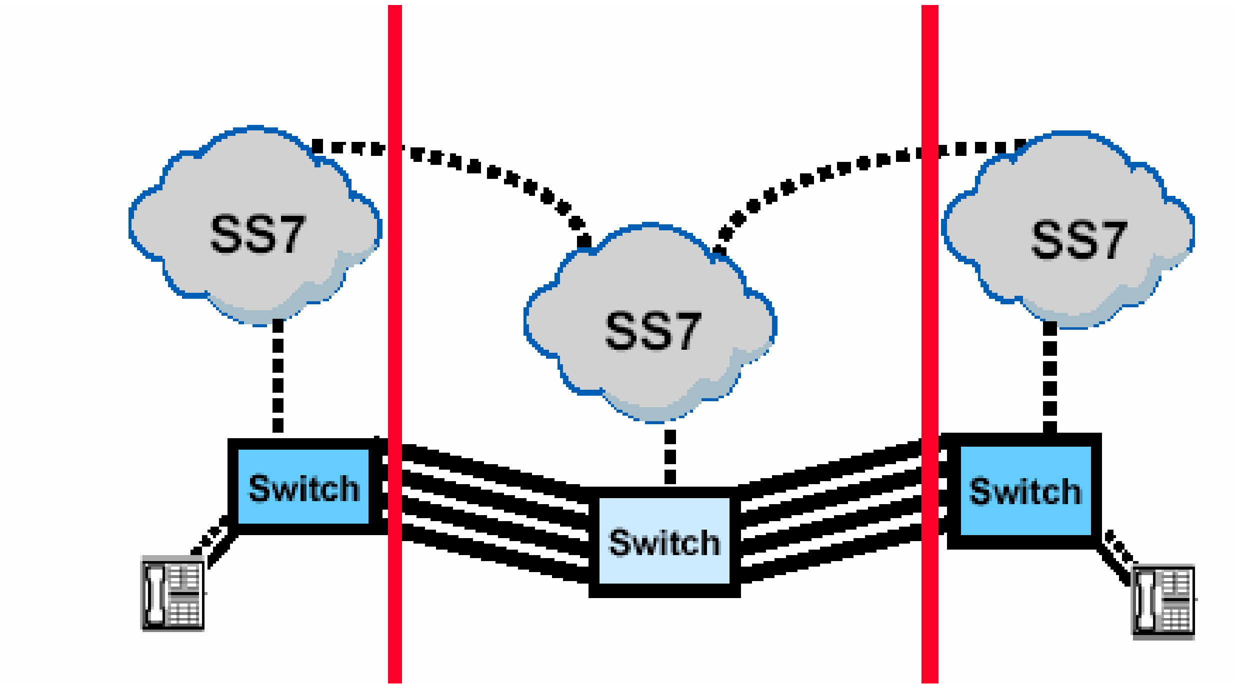
Simpler network
More efficient network
Cheaper network components
Full benefit only realized when all separate networks have fully migrated towards to the target solution

FMC –FixedMobile Convergence (3GPP and non 3GPP)
Seamless Service Continuity over different access networks (Fixed,Cellular, Wlan)
Seamless Handover between access technologies
Seamless escalation between Voice, Video and Text communication

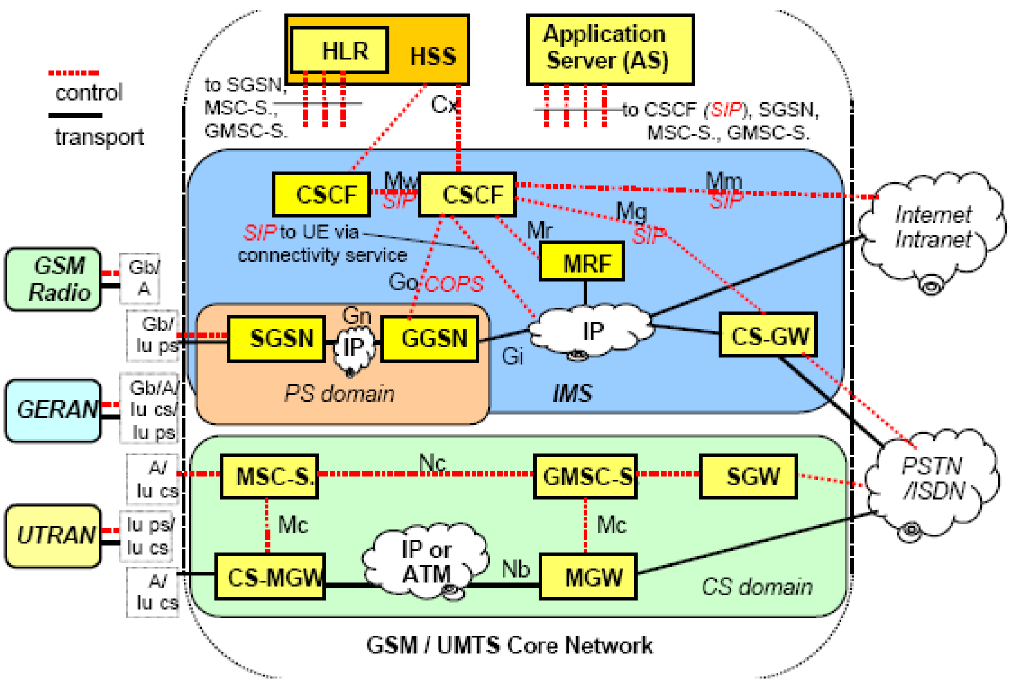
8 Evolution of network architecture

Or we could use the title from CS to IMS

Traditional telephony - Circuit switch



IMS


control
transport
HLR
HSS
Application Server (AS)
to SGSN, MSC-S., GMSC-S.
Cx
to CSCF (SIP), SGSN, MSC-S., GMSC-S.
CSCF
Mw
SIP
CSCF
Mm
SIP
Mg
SIP
Internet Intranet
SIP to UE via connectivity service
Mr
MRF
Go
COPS
GSM Radio
Gb/ A
Gb/ Iu ps
SGSN
Gn
IP
GGSN
Gi
IP
CS-GW
PS domain
IMS
GERAN
Gb/A/ Iu cs/ Iu ps
A/ Iu cs
MSC-S.
Nc
GMSC-S
SGW
PSTN /ISDN
Mc
Mc
UTRAN
Iu ps/ Iu cs
A/ Iu cs
CS-MGW
IP or ATM
Nb
MGW
CS domain
GSM / UMTS Core Network

9 Main Protocol for new implementations

Gateway control

The target of the Gateway control - to enable a simple media gateway implementation with intelligence centralized on a media gateway controller (which is also called a call agent or a Soft-switch)

Two gateway control protocols:

Media Gateway Control Protocol (MGCP) as the de facto standard
H.248/Megaco as the ITU and IETF approved standard.

MGCP/Megaco/H.248

MGCP - Media Gateway Control Protocol, IETF [Telcordia (formerly Bellcore)/Level 3/Cisco]
MGCP – control protocol that specifically addresses the control of media gateways
Megaco/H.248 (IETF, ITU) - standard that combines elements of the MGCP and the H.323, ITU (H.248)

The main features of Megaco - scaling (H.323) and multimedia conferencing (MGCP)

Media control
Media control is a form of device control used for network elements that are specialized for advanced media processing.

Media control includes instructions to play and record voice files, collect and generate tones (including DTMF touch-tones), establish N-way conferences, perform fax conversions, generate text-to-speech, and perform speech recognition

SIGTRAN

A telephone company switch transmits SS7 signals to a SG. The gateway, in turn, converts the signals into SIGTRAN packets for transmission over IP to either the next signaling gateway.

The SIGTRAN protocol is actually made up of several components (this is what is sometimes referred to as a protocol stack):

Standard IP

Common signaling transport protocol (used to ensure that the data required for signaling is delivered properly), such as the Streaming Control Transport Protocol (SCTP)

Adaptation protocol that supports "primitives" that are required by another protocols.

Bearer Independent Call Control

ISUP messages carry both call control and bearer control information, identifying the physical bearer circuit by a Circuit Identification Code (CIC). However, CIC is specific to time-division multiplexed TDM networks. BICC was developed to be interoperable with any type of bearer, such as those based on asynchronous transfer mode ATM and IP technologies, as well as TDM.

BICC separates call control and bearer connection control, transporting BICC signaling independently of bearer control signaling. The actual bearer transport used is transparent to the BICC signaling protocol - BICC has no knowledge of the specific bearer technology

The ITU announced the completion of the second set of BICC protocols (BICC Capability Set 2, or CS 2) in July 2001; these are expected to help move networks from the current model - which is based on public-switching systems - to a server-based model. The BICC deployment architecture comprises a proxy server and a media gateway to support the current services over networks based on circuit-switched, ATM, and IP technologies, including third-generation wireless.

The completion of the BICC protocols is an real and important ITU step toward broadband multimedia networks, because it will enable the seamless of circuit-switched TDM networks to high-capacity broadband multimedia networks. The 3GPP has included BICC CS 2 in the UMTS release 4. Among the future ITU-T plans for BICC are the inclusion of more advanced service support and more utilization of proxies, such as the SIP proxy.

SIP-T –SIP-I

SIP-T (SIP for telephones) is a mechanism that uses SIP to facilitate the interconnection of the PSTN with IP. SIP-T defines SIP functions that map to ISUP interconnection requirements.

This is intended to allow traditional type services to be seamlessly handled in the Internet environment. It is essential that SS7 information be available at the points of PSTN interconnection to ensure transparency of features not otherwise supported in SIP. SS7 information should be available in its entirety and without any loss to the SIP network across the PSTN-IP interface

Call Control (Session Control)

The ability of a network element to establish new calls, a “call” in the next generation network can be viewed as a session in which the session establishes either a voice conversation or, ultimately, a multimedia (audio plus video) stream.

There are two primary call control protocols unique to packet-based networks:

H.323

SIP

H.323, ITU-T

H.323 - first call control standard for multimedia networks.
Was adopted for VoIP by the ITU in 1996

H.323 is actually a set of recommendations that define how voice, data and video are transmitted over IP-based networks

The H.323 recommendation is made up of multiple call control protocols. The audio streams are transacted using the RTP/RTCP

In general, H.323 was too broad standard without sufficient efficiency. It also does not guarantee business voice quality

SIP - Session Initiation Protocol, IETF

SIP – client-server protocol, Rq from clients, Rs from servers. Participants are identified by SIP URLs. Requests can be sent through any transport protocol, such as UDP,TCP and SCTP.

SIP defines the end system to be used for the session, the communication media and media parameters, and the called party's desire to participate in the communication.

Once these are assured, SIP establishes call parameters at either end of the communication, and handles call transfer and termination.

The Session Initiation Protocol is specified in IETF Request for Comments (RFC) 2543.

Main transport protocols

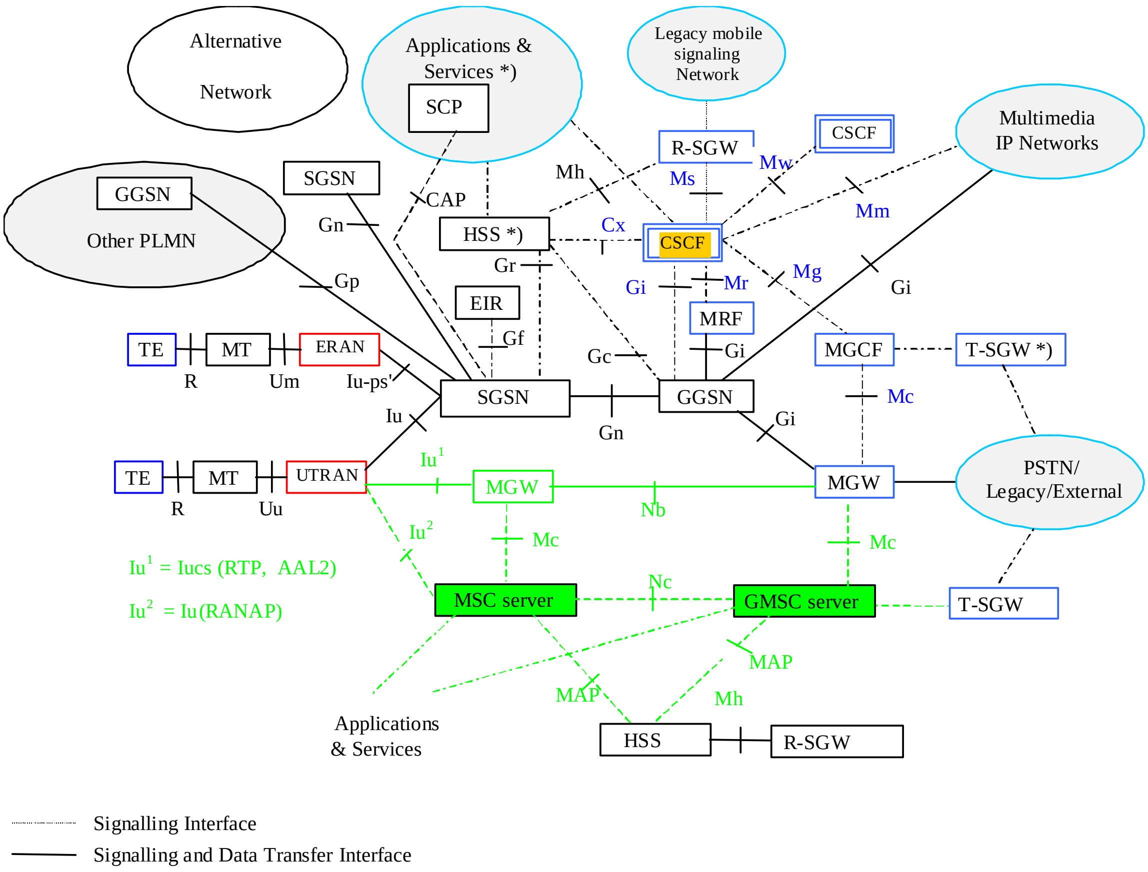
Real-Time Transport Protocol (RTP) and Real-Time Control Protocol (RTCP)
RTP - for end-to-end network transport of communications services requiring real-time data (i.e., audio and/or video).
Real-Time Control Protocol (RTCP) – for data transport monitoring
RTP and RTCP are designed to be independent of the underlying network layers (e.g., UDP/IP, MPLS, or ATM).

10 IP Multimedia SubSystem (IMS)

3GPP consortium consists of ETSI, ARIB, TTA, T1 and CWTS
Now in 2008 many networks are migrating to Rel4
UMTS R5 is an all-IP architecture with support for CS terminals.
We have already Rel 5, R6, Rel 7 currently working on Rel 8
All-IP architecture is based on GPRS (PS) with multimedia enhancements
Support for integration of intelligent services (SIP based, OSA, and CAMEL)

IMS is based on IETF protocols:
SIP is used for establishing and terminating IP communication sessions
RTP/RTCP for media transport
SDP for capability negotiation
DIAMETER for AAA
COPS for policy based QoS control
IP-SEC for inter-domain trust relations
H.248 (MEGACO) is used for gateway control

Below a General IMS topology:



IMS-Architecture Layers
CSCF-Call Session Control Function-the IP Switch
Here we have three components:
S- CSCF Serving CSCF
P- CSCF Proxy CSCF
I- CSCF Interrogating CSCF
IP-CAN –IP Connectivity Access Network
aGW- Access Gateway(RAN,SGSN,GGSN)
Media Gateway –Media Transort and Modification
MGCF - Media Gateway Control Function
MGW - Media Gateway
MRF - Media Resource Function-with two components
MRFC - Media Resource Function Controller
MRFP - Media Resource Function Processor

Service Control -Interworking with Application-Server
SIP - Session Initiation Protocol (Mandatory)
CAMEL –GSM's intelligent network approach (Optional)
OSA -Parlay Service Brokering Approach(Optional)
Repository
HSS-Home Subscriber Server
SGW-Signalling Gateway- mapping between SIP and SS7
R-SGW –Roaming Signalling Gateway (GSM MAP)
T-SGW –Transport Signalling Gateway (Telephony ISUP)

Descriptions of Interfaces:

Interfaces	**Components IMS**	**Protocol**
Gm	UE, P-CSCF	SIP
Mw	P-CSCF, I-CSCF, S-CSCF	SIP
ISC	S-CSCF, I-CSCF, AS	SIP
Cx	I-CSCF, S-CSCF, HSS	DIAMETER
Dx	I-CSCF, S-CSCF, SLF	DIAMETER
Sh	SIP AS, OSA SCS, HSS	DIAMETER
Si	IM-SSF, HSS	MAP
Dh	SIP AS, OSA, SCF, IM-SSF, HSS	DIAMETER
Mm	I-CSCF, S-CSCF, external IP network	Not Specified
Mg	MGCF -> I-CSCF	SIP
Mi	S-CSCF -> BGCF	SIP
Mj	BGCF -> MGCF	SIP
Mk	BGCF -> BGCF	SIP
Mr	S-CSCF, MRFC	SIP
Mp	MRFC, MRFP	H.248
Mn	MGCF, IM-MGW	H.248
Ut	UE, AS (SIP AS, OSA SCS, IM-SSF)	HTTP
Go	PDF, GGSN	COPS
Gq	P-CSCF, PDF	DIAMETER

Below few requirements of IMS:

Use IETF protocols (SIP, SDP) and request any additions to be standardized by IETF

Efficient use of radio interface
Signal compression

Minimum session setup time
Higher registration overhead and session based security

IPv4&IPv6 support

Network initiated de-registration and session termination

QoS support
Correlation of session and bearer establishment

Access and admission control
Policy based control

Private/Public user identity
Remote identity presentation, hiding and assertion

Hiding of network topology

Emergency services

Charging
Support for pre- and post-paid
Correlation between session and media

DTMF and early media are supported

SIP implementation for IMS

Here we will have few headers more
P-Headers are used to convey information not included in standard SIP
PATH and Service-Route

Additions to some headers
WWW-Authenticate and Authorize
VIA, Route …

Stricter routing paths:
P-CSCF to S-CSCF to I-CSCF to S-CSCF to P-CSCF

XML body used for transporting information from HSS to the SIP elements (emergency)

Specification of timer values (like: request retransmission)

More intensive use of some of SIP and SDP extensions (PRACK, UPDATE, qos, offer-answer ...)

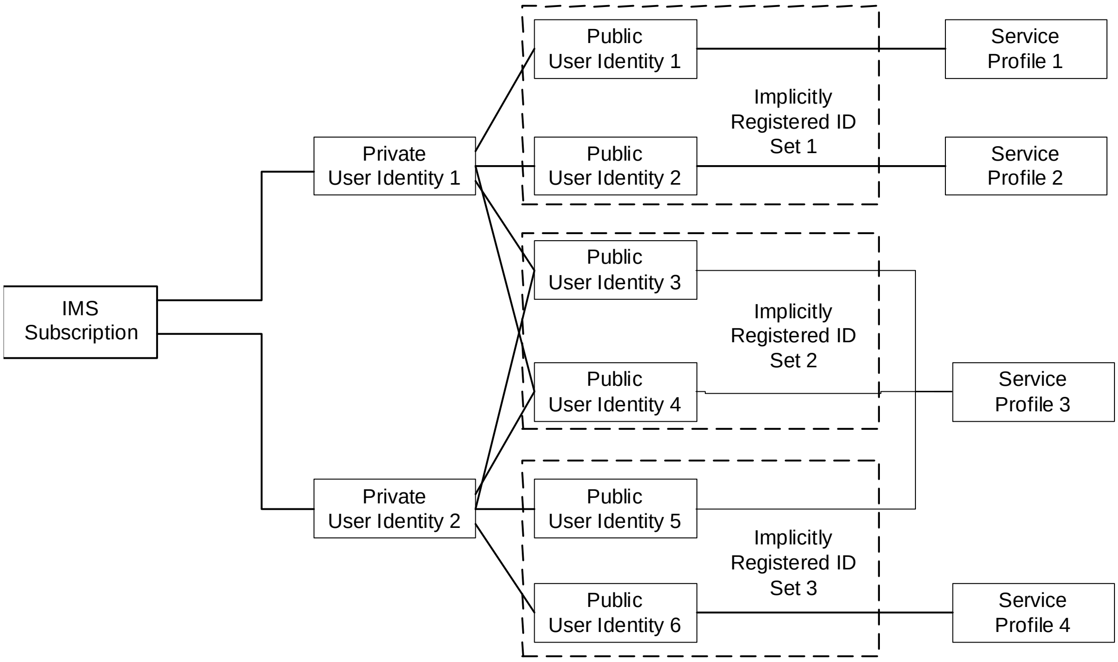
User Identify

IMPI-Private identity
Saved on ISIM (not modifiable)
Used for AAA
Issued by home provider

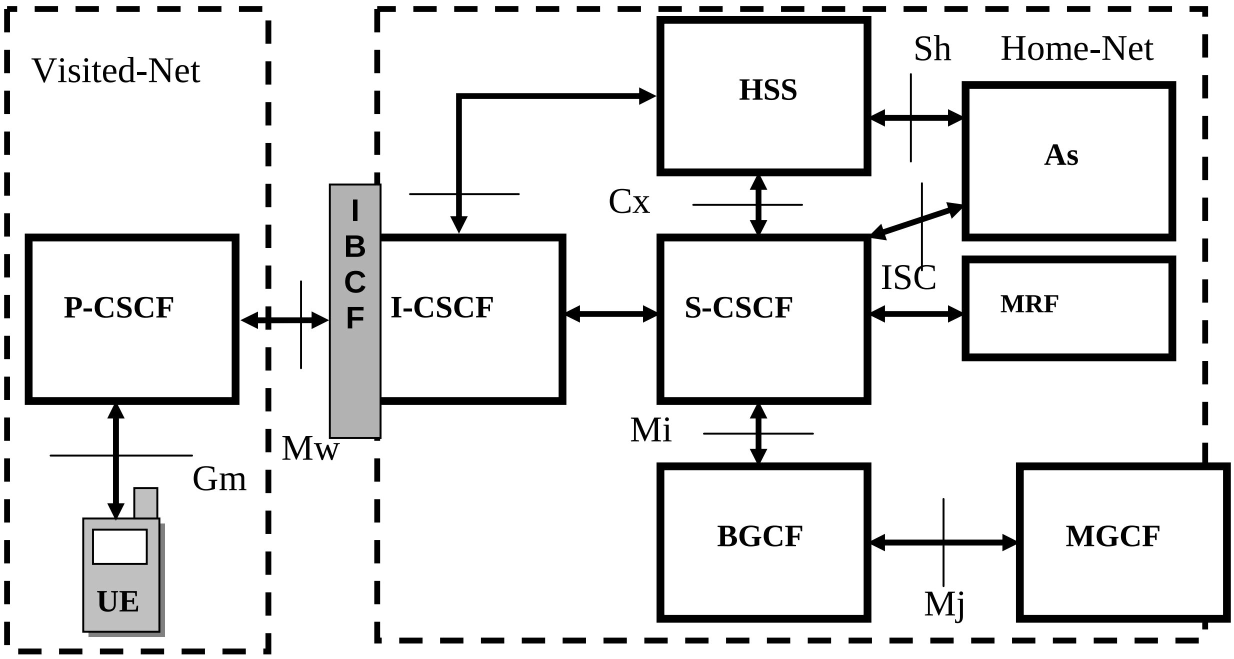
IMPU-Public identity
Normal SIP address (URI or TEL number format E164)
Identifies the user publicly
User has one or more identities
Used for routing
Can be grouped into implicit registration sets
If one of the set is registered then the others are as well

At least one is stored on ISIM
In case no ISIM is provided
Use a temporary identity derived from USIM during initial registration (derived from IMSI)
PIDs are then provided by the S-CSCF in its reply to the registration

Below the way to build a service profile:



11 IMS Components



P-CSCF- Proxy Call Session Control Function.

Here we have the first IMS contact point for the UE this can be named-outbound proxy.
This is in charge with:
Forward registration to I-CSCF
Forward requests to S-CSCF or I-CSCF
Forward replies and incoming requests to UE
Maintain security association with UE
Responsible for compression/decompression
Maintain session and registration information
Can terminate registrations or sessions if deemed necessary
Correlation between SIP and QoS
Enforce local policies
Possibly support routing to local service infrastructure
Discovered through DHCP or during GPRS PDP establishment
Generate CDRs
Emergency call handling

I-CSCF-Interrogating Call Session Control Function

The contact point within an operator
Discovered through DNS
Assign S-CSCF to a user by contacting the HSS
It's act as a THIG -Topology Hiding Inter-Network Gateway
Always on the path (RR and Service-Route) of any message leaving the network
Encrypt all entries added by the hiding network in outgoing messages looking like with release 7 this functionality has moved to IBCF
Hew are generate CDRs also.

S-CSCF - Serving Call Session Control Function

Has registrar functionality.
Acts as a SIP proxy (forward messages ...)
It is allocated to a user during registration
Always on the path of the user's SIP messages (use Service-Route and RR)
Enforces service policies based on the user's subscription profile
Collects session information for billing
Interacts with application service platform
Chose the appropriate AS based on user profile (initial filter criteria –IFC)
Forward to AS using ISC interface
Acts as user agent when required (Notifications about de-registrations and re-authentications, call termination)

UE-User Equipment

Contains the SIP user agent
Establishes a PDP context for:
Signaling
Media transport

Contains ISIM for authentication
Public and private user id
User Network address
Security algorithms and keys
At least a USIM
Correlate between session control and QoS reservation

BGCF-Breakout Gateway Control Function

Select PSTN/CS domain to forward call to
Local MGCF
Another BGCF

MGCF-Media Gateway Control Function

Gateway to PSTN networks
Translate SIP messages in appropriate PSTN signals and vice versa
Establish bearer with appropriate code
Possibly translate codec
Act as UA (but no registration required)

MRF –Media Resource Function

Here we have two components –MRFC and MRPF
Provide conferencing and announcement services
Use H.248-MEGACO (or equivalent) between the two components involved

Multimedia Resource Control Function (MRFC)
Interpret information from S-CSCF and AS
Conference booking and floor control from AS for example
Control MRPF

Multimedia Resource Processor Function (MRPF)
Establish bearers based on MRFC requests
Media mixing and distribution
Media streaming for announcements

AS –Application Server

Services include third party CC, personalized routing, PTT, presence
The services are offered by home, visited or third party provider
S-CSCF forwards requests to AS base (possible received from HSS-depends on profile)
Results of AS sent back to S-CSCF

AS can act as UA, redirect or proxy-CAMEL and OSA optional

ISC-IMS-Service Control
Based on SIP–SIMPLE **(SIP for Instant Messaging and Presence Leveraging Extensions)**
S-CSCF could add charging information
S-CSCF could add information to allow the distinction between incoming and outgoing messages

HSS-Home Subscription Server

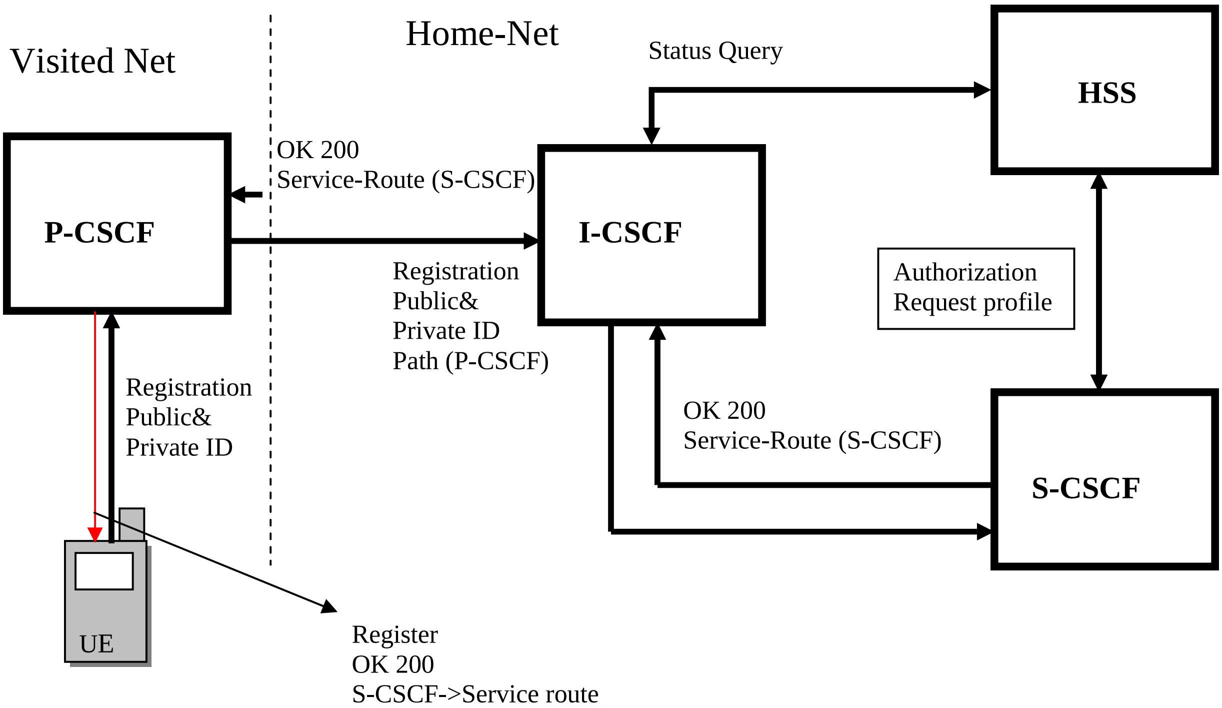
Contains user profile information indicating
Private and public identities of the user
Authentication information
Which services and medias the user is eligible for using
Filtering criteria for choosing appropriate AS
Assist I-CSCF in choosing the appropriate S-CSCF
Maintain subscription information about the user
Enforce provider policies
De-register users with invalid subscription
Connected through Cx interface to S-CSCF and I-CSCF (DIAMETER)
Connected also to AS (Sh interface)
Provide user service information
Allow multiple instances by using SLF (Subscription Location Function)
I-CSCF asks over Dx the SLF which HSS is responsible for the user

IBCF-Interconnect Border Control Function

This is an optional component
If used then replaces the I-CSCF as the entry point to the network
It is built as B2BUA- **Back-to-Back User Agent**
B2BUA-acts as a user agent to both ends of a SIP call. The B2BUA is responsible for handling all SIP signaling between both ends of the call, from call establishment to termination. Each call is tracked from beginning to end, allowing the operators of the B2BUA to offer value-added features to the call.
To SIP clients, the B2BUA acts as a User Agent server on one side and as a User Agent client on the other (back-to-back) side.
The B2BUA may provide the following functionalities: call management, network inter-working, hiding of network internals (private addresses and network topology), and codec translation between two call legs

12 Few example of IMS standard scenarios

Registration
Before the UE performs the GPRS procedure and activates a PDP context for SIP signalling**.** Here this procedure is not described.



Registration: Request handling

P-CSCF behavior(from visited net or foreign net)
UE adds private and public identity in the REGISTER message
P-CSCF adds a PATH header with its address to the REGISTER message
P-CSCF adds P-Visited-Network-Identity to the message
Discover the I-CSCF of the user using DNS

I-CSCF behavior
Determine the right S-CSCF
Ask an HSS (Cx Interface with DIAMETER)
Ask an SLF (Subscription Location Function) about which HSS to use (Dx Interface with DIAMETER)
Adds itself to the PATH list

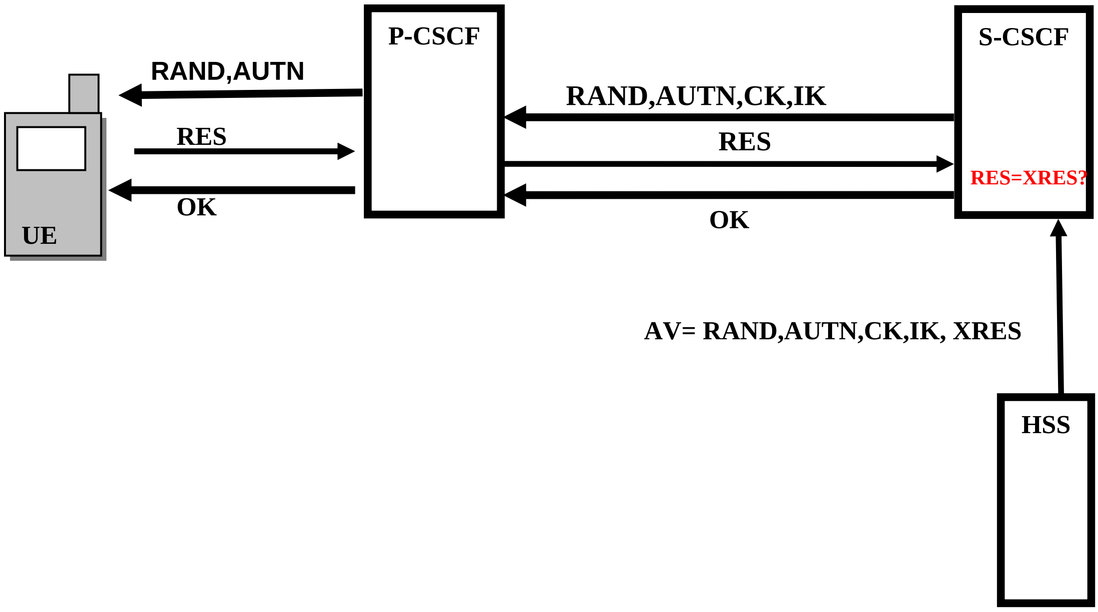
S-CSCF behavior
Download the user profile from the HSS
Save contents of PATH
Generate reply
Forward to AS if needed

Registration: Reply handling
S-CSCF behavior
Add “service-route” to reply with the address of the S-CSCF

I-CSCF behavior
If it is to stay in the path of future requests then adds itself to the “service-route” list

P-CSCF behavior
Store content of “service-route”
Store the public user identities found in the P-Associated-URI

Access Security in IMS



AV = RAND||AUTN||XRES||CK||IK

RAND = Random number
AUTN = SQN⊕AK || AMF || MAC
MAC = Message authentication code
AMF = Authentication Management Field
AK = Anonymity key

XRES = Result

CK = Cipher key = f3 (K, RAND)

IK = Integrity key = f4 (k, RAND)

AK = f5 (K, RNAD)
SQN = AK (AUTN)

XMAC = f1 (K, (SQN|RAND|AMF)

XMAC = MAC?
RES = f1 (K, RAND)

CK = f3 (K, RAND)
IK = f4 (k, RAND)

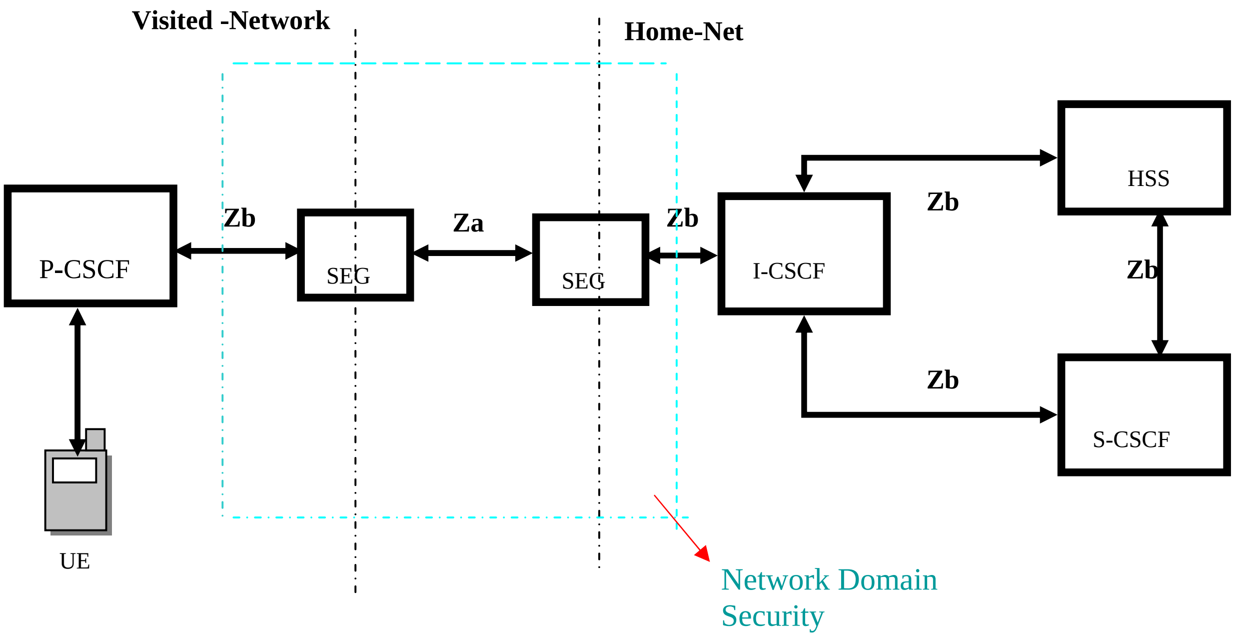
UE (ISIM) and HSS (AuC) share a secret K

Based on AKA which provides
Mutual authentication between user and network
Temporary shared key between UE and P-CSCF
Used for establishing an IPSEC tunnel between UE and P-CSCF

In case of re-registration, the P-CSCF indicates whether the registration was received in a secure manner.

Besides AKA IMS supports:
Early IMS with SIP like authentication
Supporting TISPAN authentication

Security-Network Domain Security (NDS)



NDS-is using a combinations of cryptographic security mechanisms and protocol security mechanisms applied in IP security (IPsec)
Support two interfaces
Za: IPSEC connection between different networks
Zb: IPSEC connection between components of the same network

SEG: Security Gateway
Depend on the implementation it is possible to combine SEG with I-CSCF

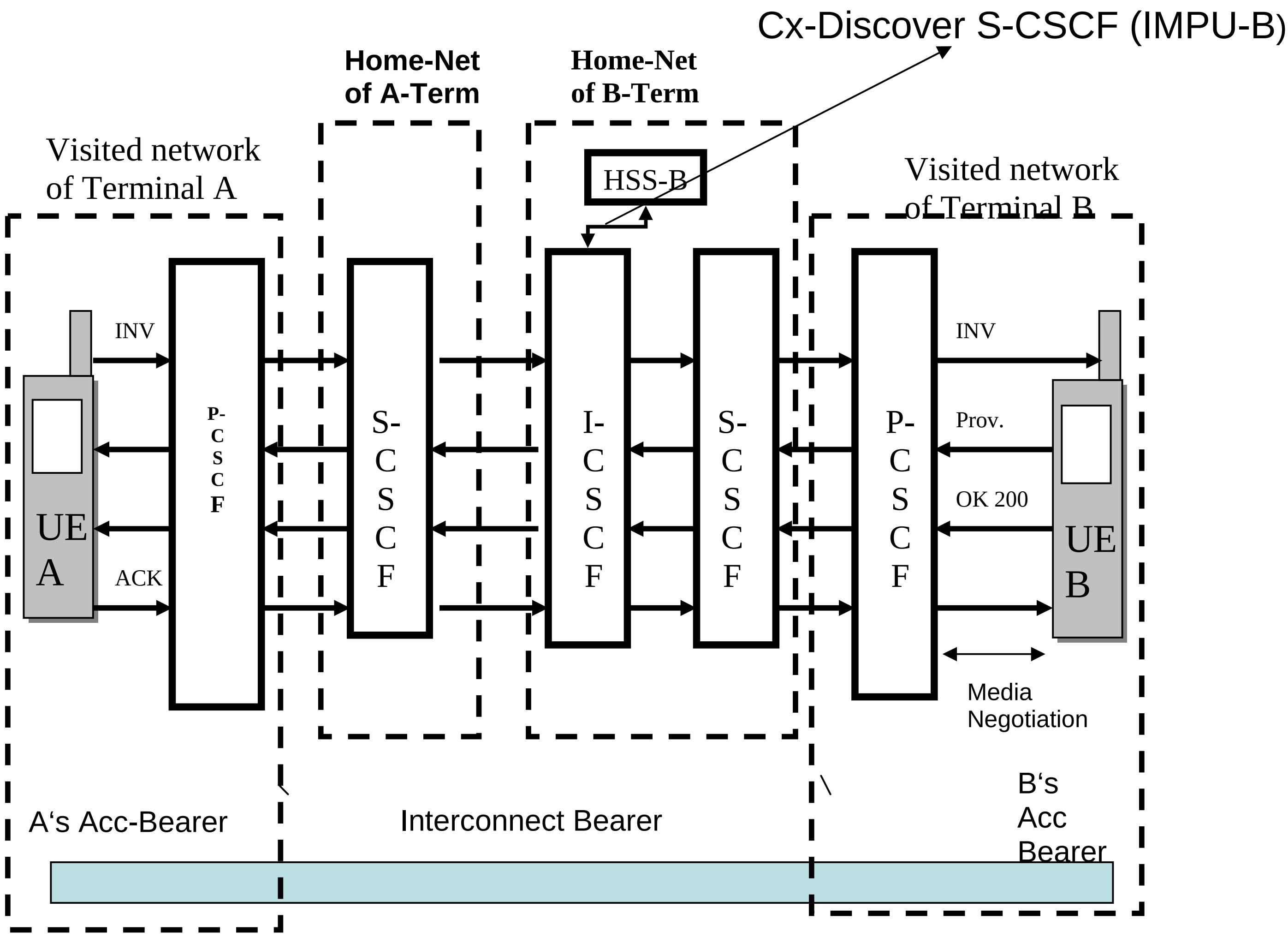
State Information

S-CSCF:
P-CSCF (PATH)
User Profile
Authentication data
Session data

P-CSCF
S-CSCF (possibly also THIG) (Service-Route)
Security association with the UE
Allows for checking the integrity and authenticity of the messages
Allows issuing a network asserted identity
P-Asserted –Identity
Used for hop-by-hop trust relations
SigComp compartments
Session data (if session termination is to be supported)
Registered public ID and the set of public IDs that were received in the P-Associated-URI header
Subscription to registration state of PID

I-CSCF
Could cache a PID(procc –id) to S-CSCF translation

IMS-VOIP Session Setup Flow



Traffic Bearer resource reservation

Cx-is Diameter
SIP-all other signaling

13 Few Important Characteristics of IMS implementation

QoS in IMS
UMTS can offer different QoS for different kinds of media
Conversational, streaming, interactive and background
Different classes offer different delay guarantees
Different PDP contexts are used for media and signaling
Sessions are allocated resources based on SDP
Use bandwidth parameter
Use local policies regarding used media

Session Termination- Network Initiated

Termination can occur due to bearer or service related events
P-CSCF or S-CSCF can decide to terminate a session
Act as UA using maintained state information
P-CSCF will inform the GGSN via “PCRF”-Policy & Charging Rules Function to terminate the bearer

Charging

Need to correlate bearer resources with IMS session

GGSN create charging information that is handed over to P-CSCF
Describe which resources are used
Need to update information based on changes in media PDP context

P-CSCF include the data as P-Charging-Vector in SIP messages

Addresses of charging collection functions are also transported in SIP (P-Charging-Function-Address)

SIGCOMP
SIP messages can become large
Long transmission delay
High bandwidth usage

SIGCOMP specifies a framework for enabling the compression and decompression of messages with various compression algorithms

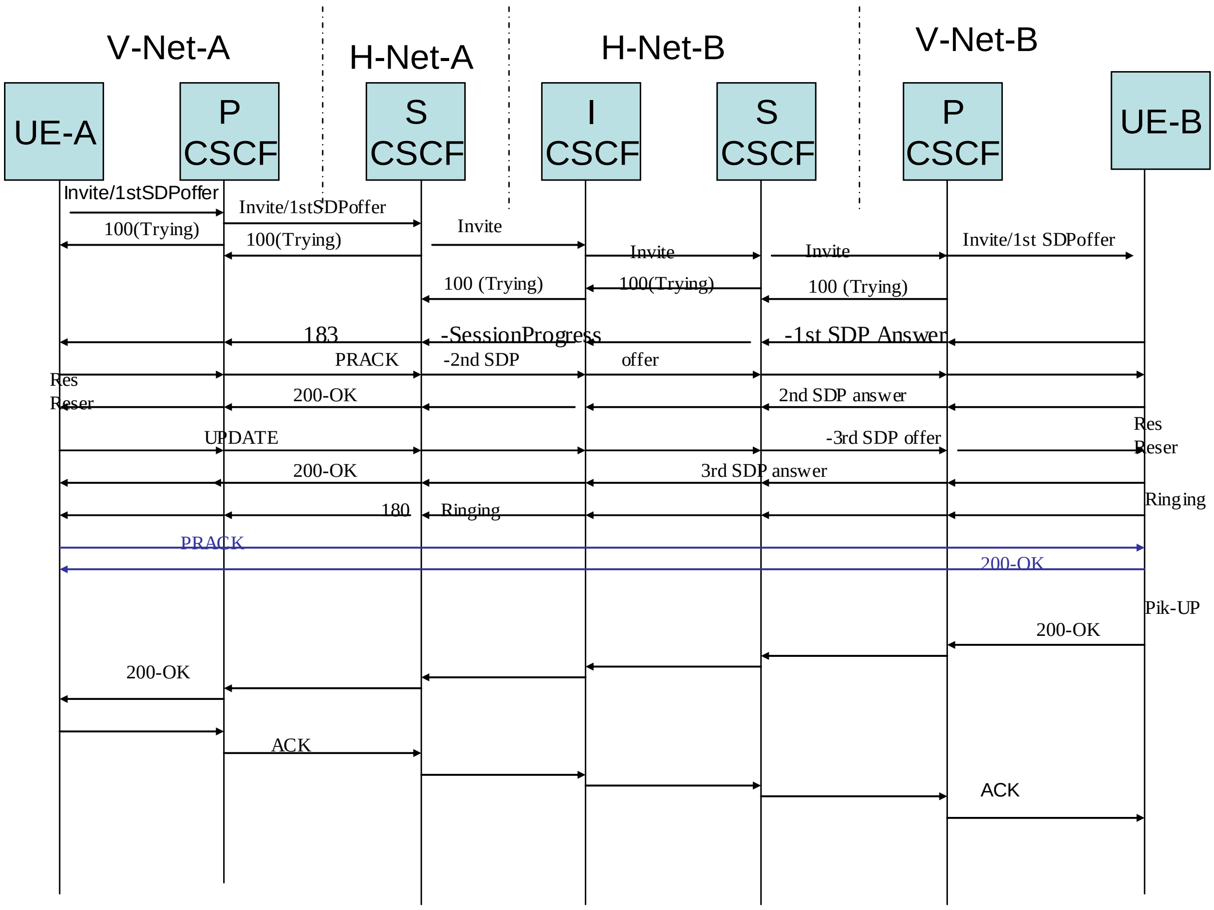
Compressor: Compresses messages and uploads the ByteCode for the corresponding decompression algorithm to the UDVM as part of the SigComp message.

Decompressor (UDVM): Uncompress messages by interpreting the corresponding ByteCode received previously

State Handler: Manages compartments with some information to use between received SigComp messages.

SigComp itself allows both parties to exchange some status information, and pointers to state to be used

14 One example of IMS Session



Obs-V-Visited &H-Home

SIP session is initiated by an "INVITE" containing proposed Media Components (SDP-Offer).
INVITE Request uses **'loose routing'**-every node looks for 'next hop towards destination'.
The received INVITE indicates the SETUP path –Via headers added by every Proxy.
The 200-OK response should fallow the same path as the INVITE->Response use **–'explicit routing'.**

Observation:
In SIP : fallowing the 200 OK the Proxies are not necessarily involved.
In IMS: Proxies which wishes to remain in the Path adds **"Record-Route"** Header to the INVITE.
The Media Steams may fallow a completely different path.

Default SIP Routing
INVITE-OK-ACK should always pass through Servers mentioned in Via Headers.
Subsequent messages are exchanged directly between the end-users **(Contact Header).**

Record-Route
Record-Route is a SIP feature that allow the P-CSCF and the S-CSCF to monitor/influence Mid-Session Signaling
Record-Route Header inserted in INVITE, is reflected in the 200 OK.

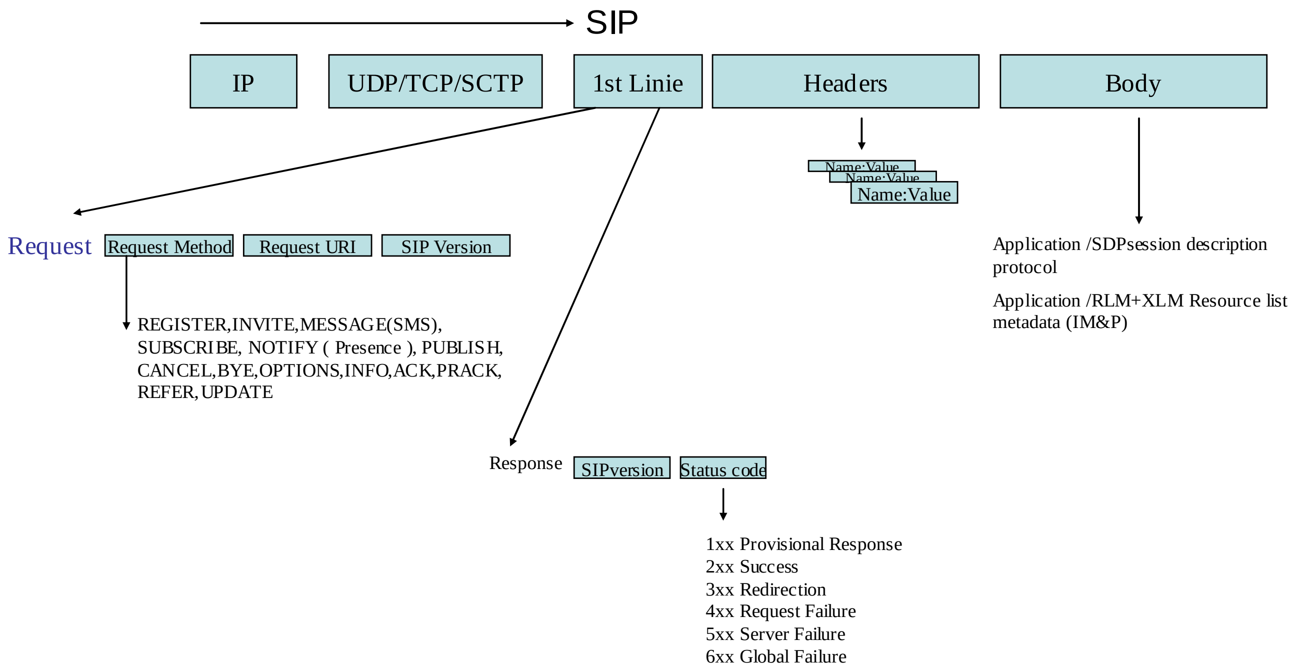
No Record-Route is necessary for:
Presence Service
Trusted Application-Server in the Home Network

Subsequent Routing –during the session

Via Headers (with multiple Branch –upon forking)
Record-Route Header –proxies that requested to remain in the Path

15 SIP Messages

SIP messages contain:
1-st Line specifies the destination and the remote operation required
Headers provide routing and application-specific information (e.g.via).
Body may contain various information formats (SDP,HTML,MMS).



Order of SIP Headers
Hop by Hop (Via)
End to End (To)
Entity Headers (Content-Length)

Body Examples
SDP –Session Description Protocol
PDIF-Presence Information Data Format (RFC-3863)
RLM-Resource List Meta Information (RFC-4662)

16 IFC-Initial Filter Criteria

IFC-Initial Filter Criteria
iFc-in the User's Profile maps SIP Trigger Events to Application Server
SIP Trigger Events –marks the reception of specific SIP messages.

S-CSCF Service Control
Received SIP messages are compared against the Filter Criteria
Upon match-the Application-Server is serialized to the call flow (via, record-route)
Transparent Service-info may be forwarded to the Application-server (e.g.IMSI)

Filter Criteria
Filter-criteria are part of the service Profile downloaded in HSS
They map SIP Events (methods, headers) to specific application server

SPT-Service Point Trigger criteria
SIP Methods: REGISTER, INVITE, SUBSCRIBE, MESSAGE
Presence, absence and contents of any header
Request-URI
Direction of the request (MO/MT)
Media Type and QoS (SDP)

17 FMC-Fixed Mobile Convergence –very important for the future implementation

Enable users to roam from CS to IP networks with no interruption of service
E.g., user in WLAN moves to 3G or vice versa
CCCF (Call Continuity Control Function) mediates between CS and IMS

IMS anchored model
Any calls generated by this user will have to be handled with in the IMS world
When a user turns on his mobile he is registered to IMS and/or CS
User is an IMS subscriber with a user profile in IMS

FMC
Seamless service continuity over different access network(Fixed, Cellular, WLAN)
Seamless escalation between Voice,Video and Text communication.

Service Convergence
Centralized Service Platform
Operate, manage and change the same Services over a range of different access technologies.

Device Convergence,
Multi-mode Device supporting multiple access technologies (fixed/cellular/wireless)

Seamless Handover between access technologies

Multi Handover Hierarchy
Homogenous Handover –within 3GPP-based on GTP
Intra RAT HO within the same Radio access Technology
Inter RAT HO between 3GPP Family technologies (e.g UTRAN to GERAN)

Heterogeneous Handover -non 3GPP-based on MIP
Heterogeneous HO between 3GPP technologies (e.g LTE->WiFi)

18 ICS-New concept from Rel 8-IMS Centralized Services

Centralized IMS Application –for all Telephony calls
Consistent User Experience-same services in All Domains
New Rule: Cs-Bearer=IMS-Service-Data-Flow->Cs-Core Telephony remains in the picture

ICS Applicability
Various Core Technologies CS-Core: GSM, UMTS, Cdma200-1x, PS-Core
Various Access Technologies :3GPP Cellular, WiFi, Wireline

ICS SCOPE Restriction
In Rel8 only AMR Speech and Video-Telephony

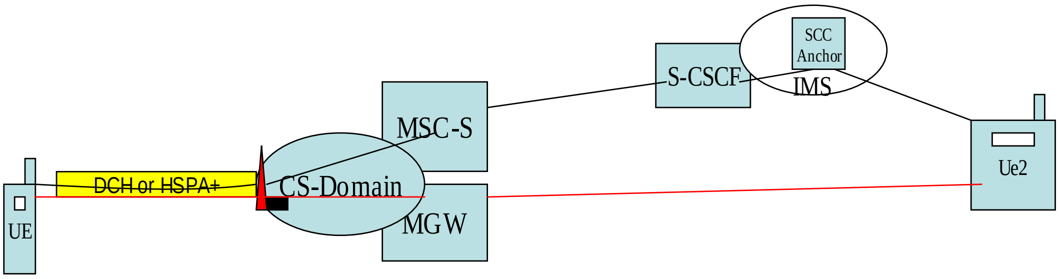
ICS Justification-1st step towards ALL-over –IMS
Most of existing PS-core Networks is still QoS-less
ICS reuses the existing Telephony infrastructure
It simplified the CS-PS convergence which becomes a Seamless CS-like Handover
Single Telephony Application –located in IMS

Call Routing is done in IMS
The Cs-Domain MSC role reduces to Inter-MSC Mobility within the 2G/3G Handover

Single Telephony Application –located in IMS
Call Routing is done in IMS
The CS-Domain MSC role reduces to Inter-MSC Mobility within the 2G/3G Network

ICS Roaming Numbers

CSRN (MSRN)	CS Roaming Number ->Mobile Terminated IMS->CS->VLR returns CSRN as a response to SRI
IMRN	IMS Roaming Number ->Mobile Originated CS->IMS->CAMEL triggered forwarding to IMRN
Dynamic ICS	PSI Public Service ID-> Mobile Originated CS->IMS->PSI received via USSD



Observation:
With black here we have CP and with red UP

19 Domain of interest

In the end of the document I will specify the domain of interest.

Based on FMC (Fixed Mobile Convergence) and ICS (IMS Centralized Services concepts) (both of them with very big importance in the future implementations) I could tell that the Tracing-System will occupies a very important place.
That will help each IMS operator to identify the problems and to improve the implementations.

I have identified this like a very important and not yet standardised domain.
I am referring here at tracing end to end between different technologies (3GPP and non 3GPP), implementations, protocols and interfaces, and to correlate end to end these different information.
I see here in this future's provocation a possibility to bring my small contribution.

20 Abbreviations

3GPP	3rd Generation Partnership Project, produces UMTS standard
AS	Application Server, in IMS
AuC	Authentication Center
BSC	Base Station Controller, controlling node in GSM RAN
CDMA	Code Division Multiple Access; each user (application) uses different "code" on the radio interface
CK	Cipher Key used for encryption between UE and RNC
CN	Core Network; in UMTS consisting of CS Domain, PS Domain and
CSCF	Call State Control Function, network element in IMS
CS	Domain Circuit-switched Domain, one of the UMTS functional groups
DCH	Dedicated Channel, one of the transport radio channels
DRNC	Drift RNC, used in Macro-diversity
DSCH	Downlink Shared Channel, one of the transport radio channels
DTCH	Dedicated Transport Channels, one of the logical radio channels
EDGE	Enhanced Data Rates for GSM Evolution, high bandwidth radio interface for GSM
EIR	Equipment Identity Register
FDD	Frequency Division Duplex; uplink and downlink use different frequencies on the radio interface
FDMA	Frequency Division Multiple Access; each user (application) uses different frequency on the radio
GERAN	GSM EDGE Radio Network
GGSN	Gateway GPRS Support Node, network element in the PS domain
GMM	GPRS Mobility Management
GMSC	Gateway MSC, network element in CS Domain, gateway to external networks
GPRS	General Packet Radio Service, 2.5 Generation system
GSM	Global System for Mobile Communications, European 2G System
GSN	Term to refer to both GGSN and/or SGSN
GTP-C	GPRS Tunnelling Protocol for the control plane between RNC and GGSN
GTP-U	GPRS Tunnelling Protocol for the user plane. Realizes PDP context between RNC and GGSN
Go	Reference Point between P-CSCF and GGSN
HLR	Home Location Register, main subscriber database in GSM and GPRS
HDSPA UMTS	High Speed Downlink Packet Access, higher data rate downlink channel for
HSCSD	High Speed Circuit Switched Data, higher data rate for GSM
HSS	Home Subscriber Server = HLR plus IMS functionality
I-CSCF	Interrogating CSCF, one role of the Call State Control Function in the IMS
IETF	Internet Engineering Task Force, responsible for Internet Standardization

IK	Integrity Key, for integrity protection of signalling messages.
IMEI	International Mobile Equipment Identity
IMS IP	Multimedia Subsystem, one of the UMTS functional groups
IMSI	International Mobile Station Identity
IMT-2000	International Mobile Telecommunications at 2000 MHz, 3G concept by ITU
IPCAN	IP Connectivity Access Network for IMS, e.g. RAN & PS-domain
ISIM	IMS SIM
ITU	International Telecommunication Union,
Iu	Reference Point between CN and RNC
LTE/SAE	Long Term Evolution and System Architecture Evolution
MBMS	Multimedia Broadcast and Multicast Service
MCC	Mobile Country Code, one constituent of PLMN identity and IMSI
MGW	Media Gateway Function
MGFC	Media Gateway Control Function
MPLS	Multiprotocol Label Switching (IETF), IP Traffic Engineering and QoS technology
MRF	Multimedia Resource Function
MS	Mobile Station (term used in GSM and GPRS)
MSC	Mobile Switching Centre, network element in CS Domain
MSIN	Mobile Station Identification Number, part of IMSI
MT	Mobile Termination, part of UE
NSAPI	Network Service Access Point Name, PDP context identifier at UE
P-CSCF	Proxy CSCF, one role of the Call State Control Function in the IMS
PDP	Packet Data Protocol
PLMN	Public Land Mobile Network -mobile telecommunication network under the control of a single operator
PSTN	Public Switched Telephone Network
P-TMSI	Temporary Mobile Station Identity, used in PS Domain QoS Quality of Service
QPSK	Quaternary Phase Shift Keying, Modulation Technique used in UMTS
R99	UMTS Release 1999
RA	Routing Area, for localizing UE in PS domain
RAB	Radio Access Bearer, Bearer extending from
RAI	Routing Area Identity, unique ID for identifying a routing area
RAN	Radio Access Network, Bearer extending from MT to SGSN
RANAP	RAN Application Protocol
RFC	"Request for Comment", Specification by IETF
S-CSCF	Serving CSCF, one role of the Call State Control Function in the IMS
SDMA	Space Division Multiple Access
SDP	Session Description Protocol, carried in SIP, encodes the actual session description
SIM	Subscriber Identity Module, part of mobile terminal in GSM
SIP	Session Initiation Protocol (IETF)
SGSN	Serving GPRS Support Node, in PS domain
SGW	Signalling Gateway Function, Node in IMS
TE	Terminal Equipment, part of UE

TMSI	Temporary Mobile Station Identity, used in CS Domain
Uu	Reference Point between UE and UTRAN
UE	User Equipment (term used in UMTS)
UMTS	Universal Mobile Terrestrial System, member of the IMT-2000 family for 3G, successor of GSM
USIM	Universal Subscriber Identity Module, part of UE
UTRAN	UMTS Radio Access Network