

Mobile Communications

Universal Mobile Telecommunications System

4. Universal Mobile Telecommunications System

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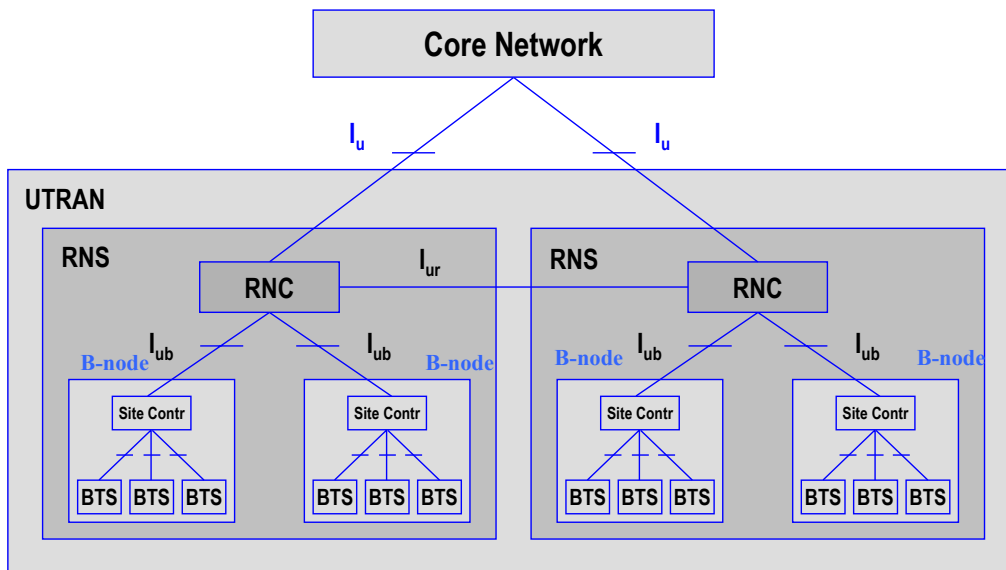


As anticipated, the definition of the UMTS Core Network architecture has been one of the controversial and dynamic aspects arisen in the standardisation process. This attitude was mostly due to the range of integration possibilities and the rapid evolution of data services. The latter is progressing in the Information technology context and will, very soon take-off in mobile systems as well, thanks to the key role played by Internet.

This section depicts the major trends in that respect, by highlighting also the side evolutionary trends, that can be ascribed to:

- IP innovation and
- the role played by ATM at switching and transport level.

UTRAN Architecture



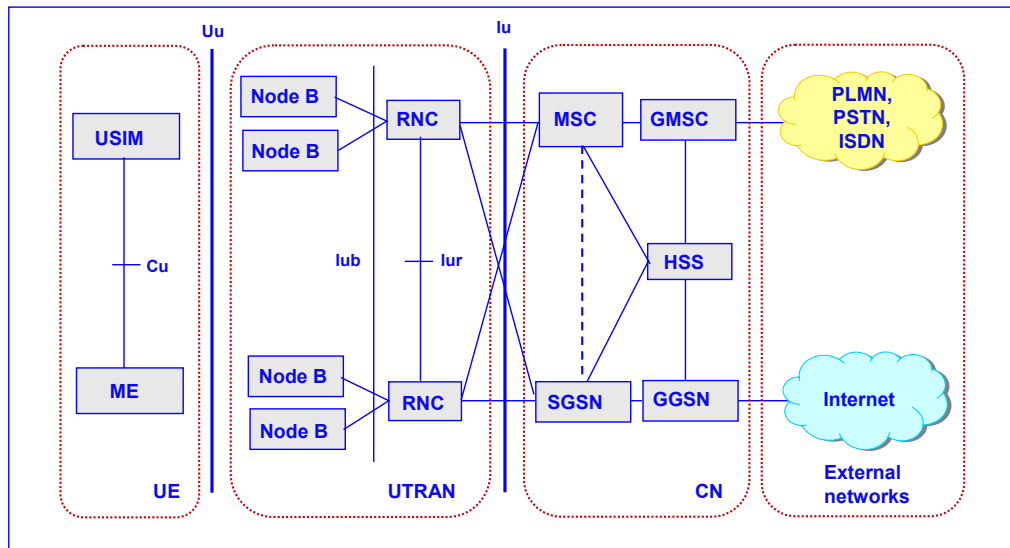
The Radio Access Network (RAN) has already been considered as for the functional allocation aspects. The depicted architecture is that adopted by ETSI for the UMTS Terrestrial Radio Access Network - UTRAN.

The Core Network architecture poses several problems that are tied both to the definition of the internal mobile-driven functions and to the integration issues arising with the other “external” networks, namely PSTN, ISDN and IP.

The section highlights the short-medium term phases for the development of the UMTS architecture, namely the release 99 and the *all IP* medium term alternative.

UMTS Release 99 is heavily based on the GSM/GPRS solution. The Core Network will be split into two sections, the circuit-switched and the packet-switched sections. Further developments beyond R99 are described as well.

Network elements and interfaces of the PLMN



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The specified interfaces include those towards the data bases for thre mobility management. In general, the UTMN network adopts the scheme followed by the GSM but the functional entities are significantly different.

The UTMN Radio Access Network (UTRAN) is composed by:

- Node B converting the radio data flow into the flow typical of the fixed network. Moreover, it partially controls the radio resources.
- Radio Network Controller (RNC) is the entity implementing all the radio control functions. It definitely hide the radio characteristics to the Core Network (CN).

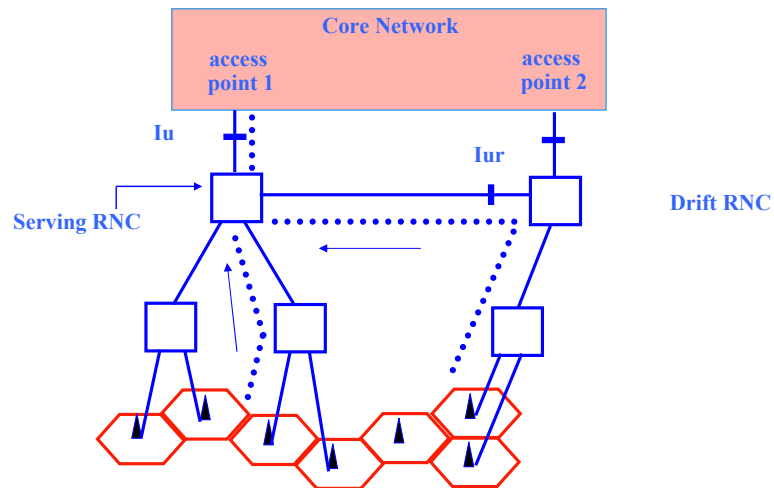
As for the GSM, the Mobile Equipment (ME) is functionally distinct with respect to the UTMN Subscriber Identity Module (USIM). The latter is strictly associated to the user and contains all his sensitive data.

The Core Network (CN) is split in two functional segments devoted respectively to the circuit end racket switching. The former includes the MSC/VLR system with GSM-like functions. The latter contains the Serving GPRS Support Node (SGSN) already seen for the GPRS. Each segment is in charge of dealing with the interface functions towards the respective networks (circuit and racket networks) through the gateways. This dual attitude impose that Iu is in practice composed by two interfaces.

The overall architecture reflects the need of maximising the consistency with the GSM system by ensuring at the same time the double control of the circuit and packet segments.

The Iur interface recovers all the function necessary to manage the upper macro diversity:that arising between the B-nodes belonging to different RNCs.

The RNC roles



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The generic RNC node performs two functions towards the UE, in function of the status of the connection.

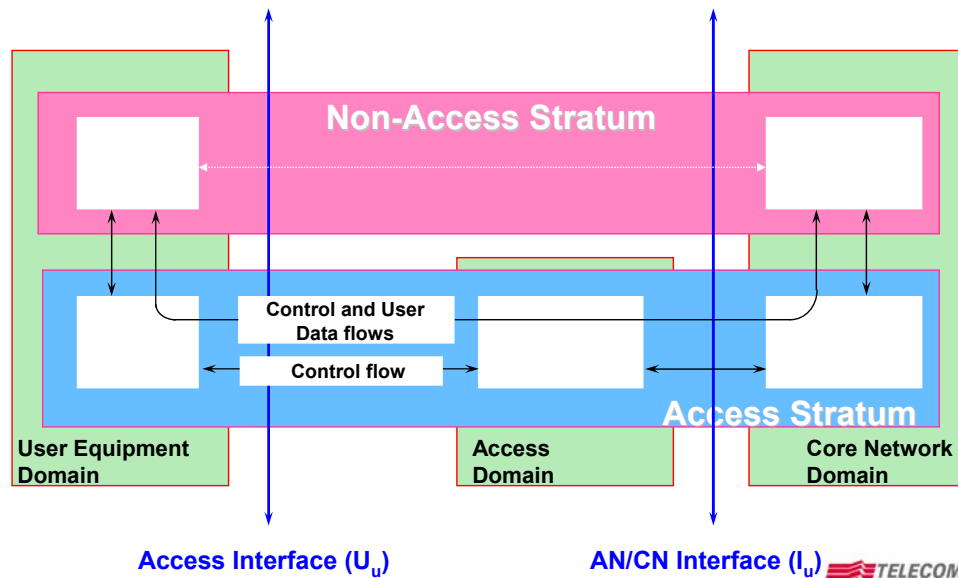
- The Serving RNC terminates the Iu interface towards/from the core network as for the data transport is concerned, as well as for the signalling needed between the UE and the CN
- The Drift RNC transports in a transparent way the data between the node b and the Serving RNC.

Independently from the role played with respect to the macro diversity mechanism, the generic RNC node implements all the control functions needed for the traffic and cell management.

The most important are:

- The open loop power control
- The execution of the call acceptance policies
- The code management
- The control of the cell congestion.

UMTS Access Protocol Framework



Within the Access/Non-Access configuration, the Iu interface is covering the signalling functional relations arising between the Access and the Core Network.

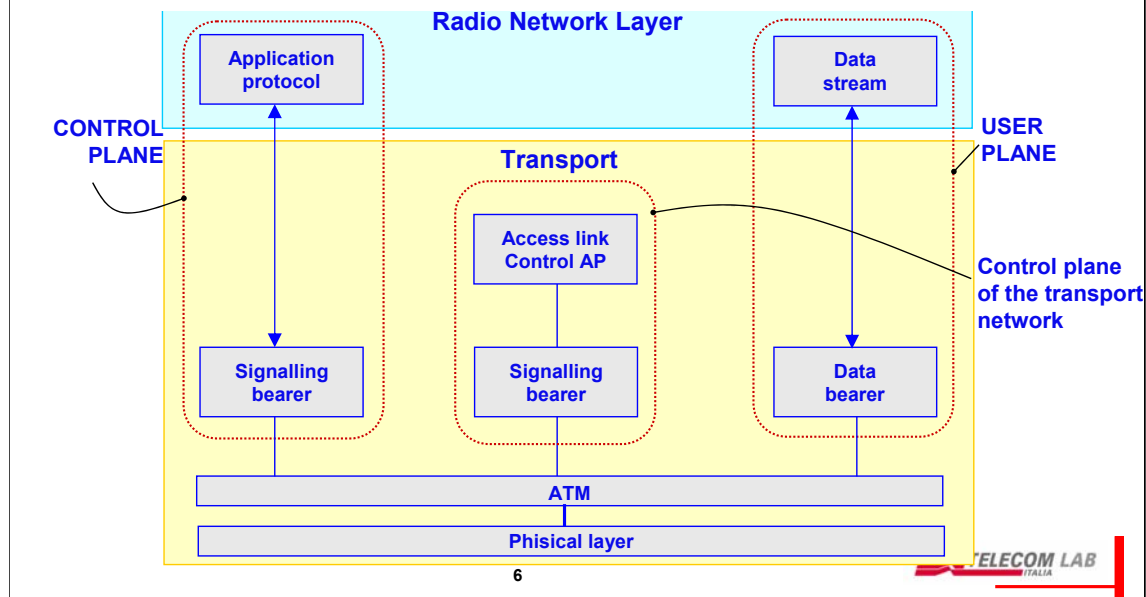
Iu is also in charge of transporting transparently the messages related to telecommunication procedures involving directly the MS and the Core Network.

In particular, Iu supplies the necessary capabilities needed for:

- the RAN procedures triggered by the Core Network such as paging notification; the paging messages are transferred by the RAN in a non-assured mode, hence their correctness is recovered through retransmission procedures not affecting the access part
- the mobile specific signalling management, through dedicated SAPs
- the Call Control and Mobility Management signalling through dedicated SAPs
- the control messages requiring the set-up of bearer and transport capabilities in the RAN
- the control messages to be exchanged for the streamlining process
- the inter-Radio Network Subsystems handover control.

All the above functions are performed through the Iu interface, where a clear separation is envisaged between the transport role and the signalling layer. Only the latter has to be defined. In fact, different transport solutions for the Iu interface may be adopted.

UTRAN interfaces – a generic protocol scheme



The UTRAN protocol scheme is that sketched in the figure.

Radio Network Bearer status controls all the interactions concerning the specific UMTS functions while the Transport Network Layer stratus is in charge of the UTRAN- internal functions.

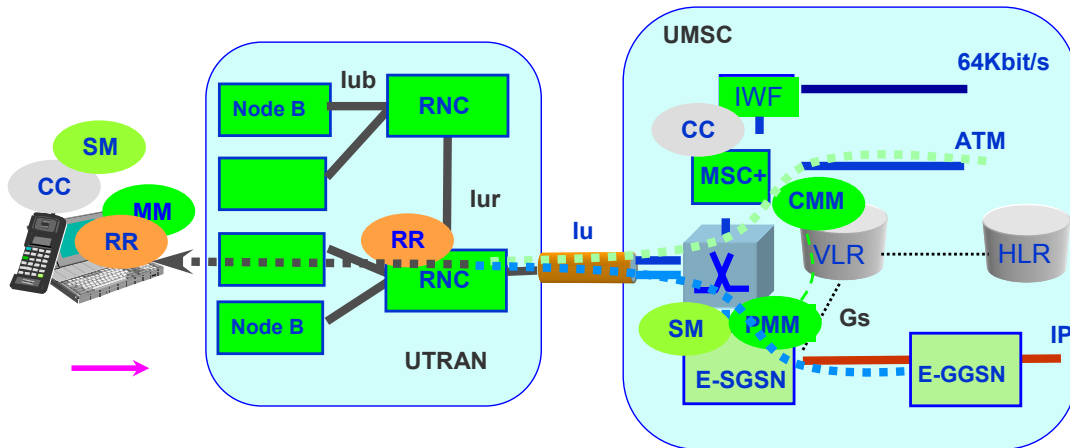
The scheme is valid for all the UTRAN interfaces (Iur; Iub; IuCS; IuPS).

A transport control has been introduced. It acts through the Access Link Control Application Part protocols that rely on specific Signalling Bearers. This stratus is in charge of de-coupling the Application Protocol from the bearers used for the data bearers of the user plane. This atratus can act also in the evolutionary term. In fact, the transport technique used for the user data can be modified with no need to modify the control protocols.

All the transport are based on the ATM technique that results to be, at least for the early UMTS implementations, the unifying technique for the access segment.

The protocol elements of the UTRAN are considered in greater detail in a specific extension.

UMTS- release '99 – a separation/integration scheme



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The specification of the UMTS-*release 99* is based on a single UMTS node, the UMTS Mobile Switching Centre (UMSC) which is able to recover both circuit and packet services.

The relevant functional entities are represented in the figure. The solution is based on:

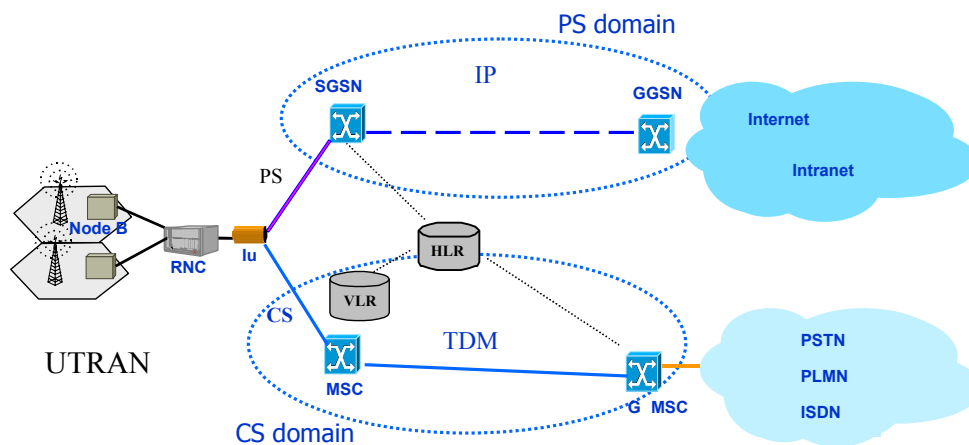
- a GSM-enhanced part for circuit-switched services where both ISDN and ATM/AAL2 schemes are adopted; Call Control and Circuit Mobility Management are directly derived from the GSM solutions
- a GPRS-based part for packet-switched data services, where a new Session Management function is specifically implemented; the enhanced GPRS Serving Node is directly using IP protocols towards data networks; a Packet Mobility Management function is conceptually sharing the same mobility control principles used in GSM.

Radio Resource Management functions are completely recovered within the Access Part.

A double backbone raises some problems that should be overcome by the subsequent version of the UMTS Core Network:

- Double planning and deployment processes
- High O&M costs
- Non efficient use of the transport resources
- Non integrated management of multimedia services (no single end point).

UMTS R99



Based on the evolution of GSM CN.

PS domain and CS domain are the evolution of, respectively, GPRS and GSM networks.

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UMTS R99 introduces an innovative radio access network, named UTRAN and based on the WCDMA radio technology.

The Core Network is based on a moderate evolution of both the Circuit Switched and GPRS GSM platforms.

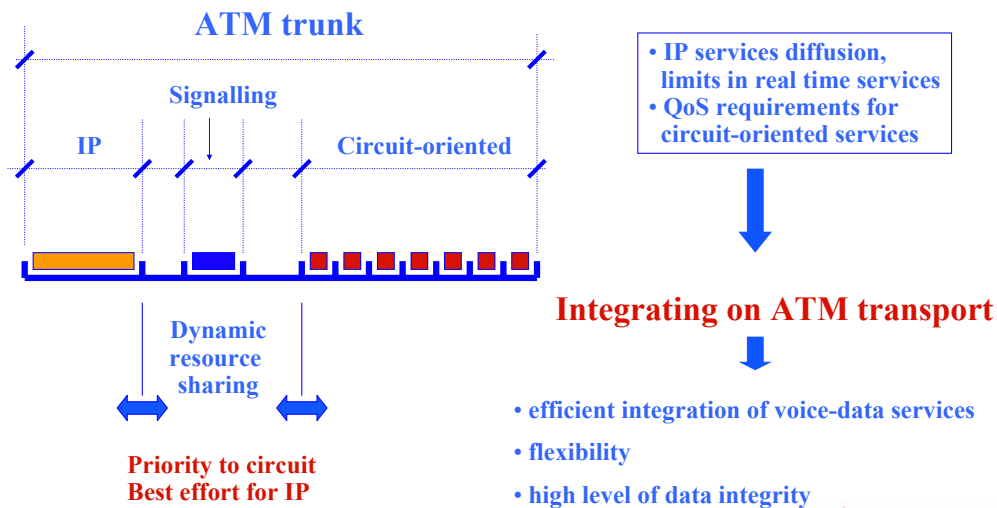
Voice and data traffic, as well as all the radio signalling protocols, are carried transparently from the MS to the RNC. Separation of PS and CS services occurs only at the Iu interfaces, where Iu_PS and Iu_CS protocol stacks have been defined to carry the relevant signalling and data towards the proper Core Network domain.

CS domain is mainly used to provide traditional telephony services and CS data. PS domain will offer packet transport for basic Internet application (Web browsing, streaming, gaming) with very poor QoS support, at least in a first stage.

Support of multimedia application is possible either on the CS or the PS domain, via respectively H.324M or H.323 mobile terminals. In both cases, multimedia (MM) terminals communicate end-to-end through the transmission of information streams that are classified into audio, video, data, communication control and call control. Thus, the UMTS network is transparent to the video, audio and data multimedia streams. However, when MM terminals of different standards communicate, the network has to connect a gateway for protocol and possible video/audio codec conversion.

Common MM applications for H.324M and H.323 mobile usage are end-user to end-user audio-visual communication for private or professional needs, but also information services that can be streamed i.e. one-directional video to a mobile terminal.

Packet-circuit mode: sharing ATM resources in the access segment



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The growing interest for IP services is more and more requiring the integration of the mobile context with Internet and Intranet configurations. Nevertheless, very likely, IP-based solutions will not be able in the short term to deal with any kind of real-time service.

At the same time, basic telephone services will presumably maintain their main role in the mobile context for the next years. This is why in the UMTS release '99, packet-oriented services and circuit-oriented services will co-exist.

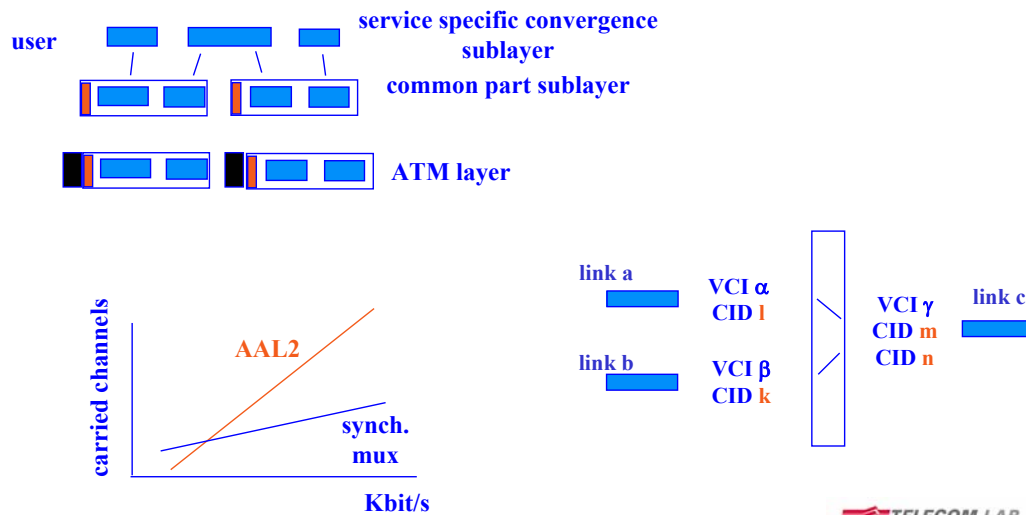
Consequently, in the access and possibly in the core network, circuit and packet services will use a common switching platform (this fulfils a basic need for efficiency in the fixed network). The needed resources should be assigned both on demand and on semi-permanent basis.

As said, ATM is the answer to these requirements; ATM is in fact supplying a flexible resource allocation mechanism and a common switching and transport medium.

The above assumptions leads to a possible shared use of any single ATM trunk; circuit-oriented services will use ATM/AAL2 virtual circuits (ensuring the needed delay requirements) and packet-oriented services will be based on a connectionless transport (IP over ATM).

The solution in figure implies common control procedures between circuit and packet modes.

ATM Adaptation Layer 2



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The use of the ATM technique in the network, poses some problems due to the substantial difference between the typical size (bits) of the radio interface blocks of information and the equivalent size of the ATM cell.

To reach a high efficiency, many radio blocks should be grouped to fill a single ATM cell: this would cause an unacceptable delay mainly for the delay-sensitive services like voice. In fact, typical sizes for radio frames are around 100 bits while the ATM payload is 48 bytes.

The trade-off between the contradictory requirements arising between efficiency and delay could be reached by adopting the ATM Adaptation Layer 2 (AAL2), specified by the ATM Forum and ITU GS13. This level of adaptation allows the multiplexing of different user connections on a single ATM connection (Virtual Call). The solution may introduce significant advantages also with respect to other synchronous multiplexing solutions.

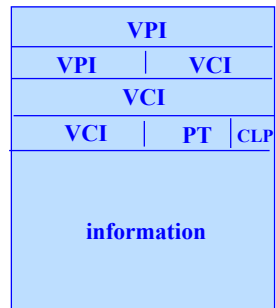
The two sub-layers of AAL2 are in this case used for:

- segmentation of the original flows (Service Specific Convergence Sub-layer)
- their multiplexing in the ATM cell, after the insertion of the additional addressing information that will be used to discriminate the single point-to-point flows (Common Part Sub-layer).

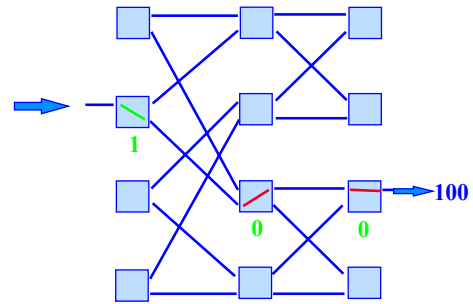
The header of a single block in the cell must identify the relevant channel, hence the related origin-destination relation. The Channel Identifier (CID) field is playing this role for the involved AAL2-layer switching functions. Also the block length is specified by the block header. The CID information is used in the transit nodes to construct the correct routing of different connections grouped under the same Virtual Call in the previous routing step.

ATM technique

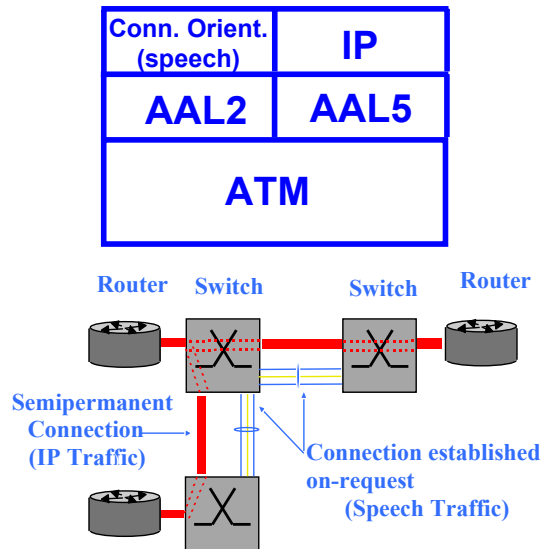
ATM cell



Binary network
(baseline)



Transport - Switching in the Core Network



- Dynamic on-request resource allocation between CO and CL links (requested QoS)
- Easy integration with Internet context
- Future-proof: ability to cope with any kind of transport and switching/routing evolution (ATM/IP)
- Flexibility: ability to handle any kind of service sharing the available bandwidth

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The previous ATM-integrated solutions can be realised by exploiting in a more extensive way the ATM flexibility: the available ATM Adaptation Layers can be used to cope with the characteristics of related traffics. AAL2 and AAL5 are the natural candidates for circuit and packet switching respectively.

The approach seems to be reasonable considering that in any case:

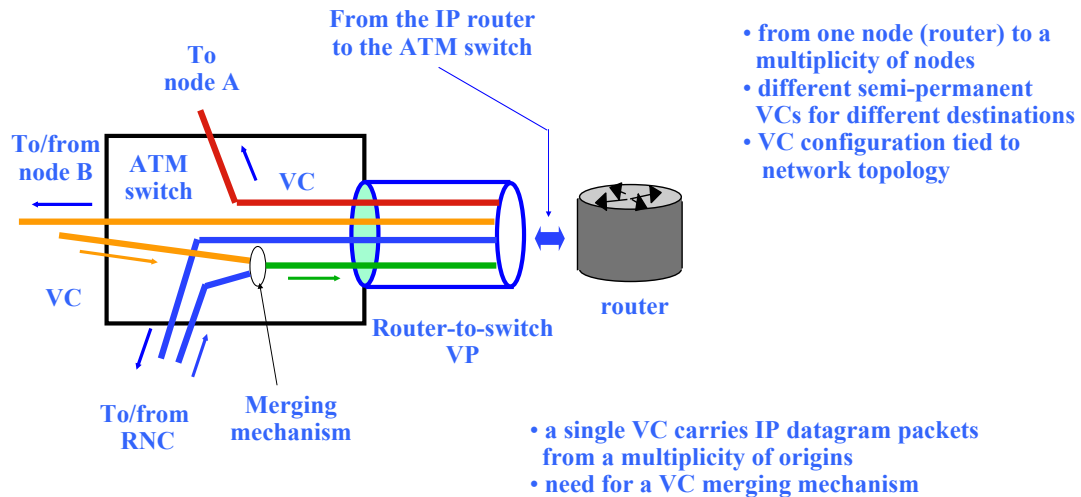
- Third Generation systems need to support efficiently integrated voice and data services
- ATM technology through its specific adaptations seems to have the right answers to the flexibility requirements induced by this integration perspective
- IP has entered the TLC market, as operators and manufacturers foresee a new business in offering Internet access through mobile and fixed telecommunication networks

IP-based technology is not ready yet to support efficiently real time services with a pure IP network and this requirement can (at least partially) be fulfilled through ATM.

Moreover, ATM can:

- emulate PCM-based voice circuits as in Second Generation network (allowing for this reasons backward compatibility)
- support the IP protocols
- offer a future-proof transport technology; in fact the proposed approach, allows UMTS networks to face efficiently the evolving transport requirements coming from data services
- support new signalling scheme for UMTS and backward compatibility towards the GSM signalling scheme (or its evolution).

Using ATM VPs and VCs in the MSC



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To send an IP datagram packet to a given node, the IP layer passes the datagram to AAL 5 and uses the Multi-protocol Encapsulation over the ATM Adaptation Layer. The datagram is fragmented in multiple AAL PDUs that are passed to the ATM layer over a specific ATM VP/VC.

The routing of data traffic is done by making use of the IP network protocol.

Any router is connected with a single Virtual Path to the relevant ATM switch.

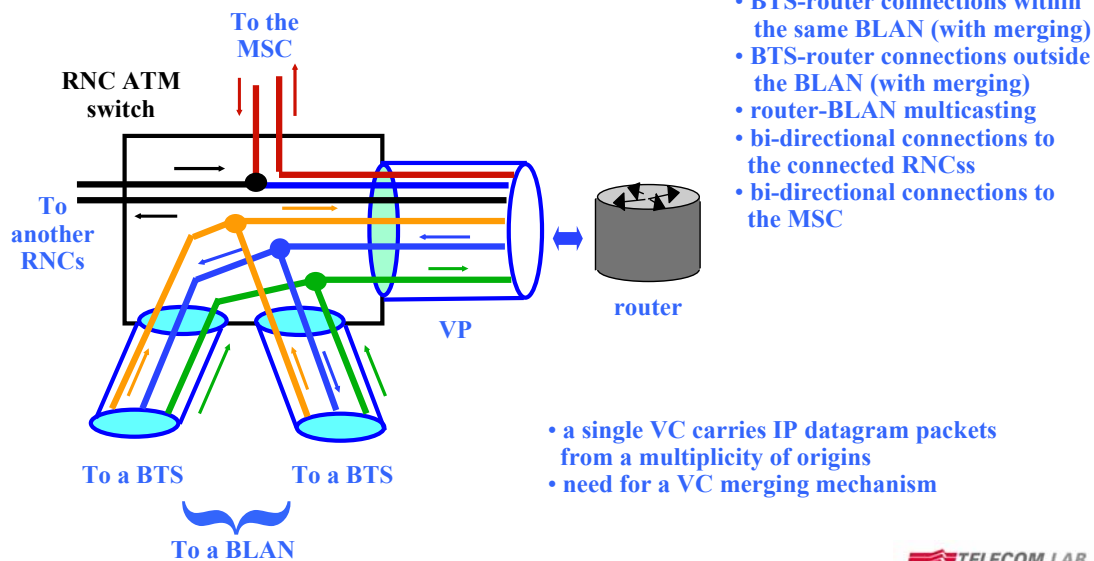
A single Virtual Call is devoted to each node topologically connected to the router.

Each VC, transports IP datagram packets that can, in general, belong to a variety of origins. As a consequence, a merging mechanism is necessary in order to avoid that ATM cells belonging to different IP packets are interleaved to each other. The alternative is to create as many VCs as the origin points are.

The VC merging mechanism uses the “end-of packet” marker (at AAL5 level) to delimit the border between consecutive packets.

A waiting mechanism avoids to start transmission of a new IP packet until the last ATM cell carrying the previous packet has been delivered.

Use of ATM VPs and VCs in the RNC



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This figure shows a possible arrangement for the RNC semi-permanent connections. Still merging mechanisms are needed.

The BTS Local Area Network (BLAN) represents the granularity according to which a mobile termination is localised according to mobile IP mechanisms. It must be possible to address the set of BTSs belonging to the same BLAN.

The above solutions are only examples, showing how the problem of semi-permanent connections for the IP-packet network can be solved when a complete IP/ATM integration is assumed for third generation mobile.

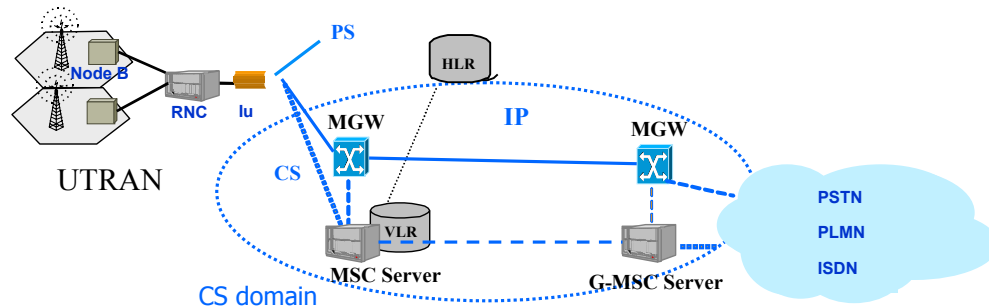
As for the transport functions, the RNC is similar to the MSC. Concerning the signalling functions, it does not contain CC and MM functions.

Evolving the circuit segment (R4)

MSC Server: call control and mobility control parts of a GSM/UMTS MSC R99.

MGW enables usage of any transport technology within the core network.

All traffic over IP but Call Control and Service Control for traditional voice services still based on the Circuit Switched concept



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In Release 4, CS domain separation of control and transport functions is a first evolution towards an IP based Core network.

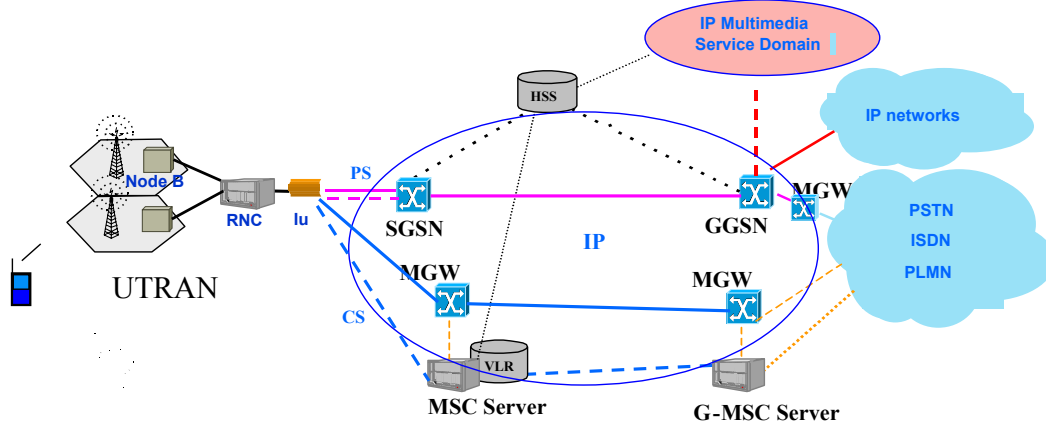
Unlike traditional networks founded on vertical integration, where each node in the core network includes transport, control and application functionality, the R4 CN is designed on a horizontally integrated network basis, where the functionality is split into a control layer and a connectivity layer.

In this architecture, a connectivity layer is introduced at the bottom for transporting user data. The key element in this layer is the Media Gateway with a generic packet switching core that is open to any type of underlying transport network (IP, ATM or STM). Servers in the control layer control the MGW in the connectivity layer using H.248 protocol. This core network provides the possibility of performing trans-coding at the edge of the UMTS network resulting in further transmission savings.

When a call is originated in the UMTS network, the MSC-server will typically point out the Media Gateway, which is closest to the end-destination. Between this Media Gateway and the originating Media Gateway speech traffic will be transported in coded form (Adaptive Multi Rate). The bandwidth reduction for this transmission path, which can naturally be quite long, can be very high.

There's already a possibility for an all-IP Core Network Transport with QoS mechanisms (IPv6, MPLS, DiffServ) as common carrier for all domains (CS and PS traffic, Signalling). In order to ensure proper support of Quality of Service (QoS) for delay sensitive traffic in a common IP core network, MPLS paths end-to-end between the nodes of the core network (e.g. MGW to MGW) can be utilised.

UMTS R5/R6



IP Multimedia Subsystem enabling SIP based multimedia services

IMS makes use of the PS domain for the transport of signaling and user data

A SIP client is included in the terminal

The PS subsystem reaches full QoS; conversational services can be supported

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The main step towards an all IP based CN is achieved in Rel5 with the specification of the IP Multimedia CN subsystem (IMS), which comprises all CN elements for provision of multimedia services.

IP MultiMedia Subsystem runs a new SIP-based Multimedia Call Control to deploy advanced multimedia applications in addition to voice and data services. With IMS not only Voice over IP is offered with true real time support; IMS offers an architecture for real time multimedia applications such as videoconference and a mix of voice/video/data services using the IP protocol.

In particular, the IMS makes use of the IP PS domain for the transport of signalling and user data. The PS domain consists of the SGSN and the GGSN nodes, which are responsible for all end-to-end IP bearers be that for best effort traffic or for real time Voice over IP. The PS domain maintains the connectivity service while the terminal moves and hides these moves from the IP multimedia subsystem.

The separation of the call control from the bearer level allows (in principle) the IMS to operate on any IP access network other than the PS domain (e.g. fixed).

The IP multimedia subsystem is independent of the CS domain although some network elements may be common with the CS domain. This means that it is not necessary to deploy a CS domain in order to support an IP multimedia subsystem based network.

The IMS high level requirements are:

- Negotiation of QoS for IP multimedia sessions
 - IP multimedia sessions able to support a variety of different media types
 - Inter-working with PSTN, ISDN and Internet
 - Support of basic voice calls between IMS users and CS users
- Roaming, enabling users to access IP multimedia services provisioned by both Home and Serving networks

The IP Multimedia domain

The diagram illustrates the IP Multimedia domain architecture, showing the integration of legacy and IP networks. The architecture is divided into two main sections: the left side represents the legacy network, and the right side represents the IP Multimedia domain.

Legacy Network (Left):

- TE (Terminal Equipment):** The user's device.
- MT (Mobile Terminal):** The mobile terminal.
- UTRAN (UMTS Terrestrial Radio Access Network):** The radio access network.
- Iu:** The interface between UTRAN and the core network.
- GPRS (General Packet Radio Service):** The core network technology.

IP Multimedia Domain (Right):

- Applications & Services:** The services provided by the IP Multimedia domain.
- Legacy mobile signaling Network:** The legacy signaling network.
- Multimedia IP Networks:** The IP Multimedia domain network.
- signaling GW (Gateway):** The gateway between the legacy signaling network and the IP Multimedia domain.
- media GW (Gateway):** The gateway between the GPRS network and the IP Multimedia domain.
- PSTN/Legacy/External:** The external network (Public Switched Telephone Network).

Interfaces:

- Signalling Interface:** Represented by a dotted line.
- Signalling and Data Transfer Interface:** Represented by a solid line.

The diagram shows the flow of signaling and data between these components, highlighting the integration of the legacy network with the IP Multimedia domain.

In addition IMS shall support:

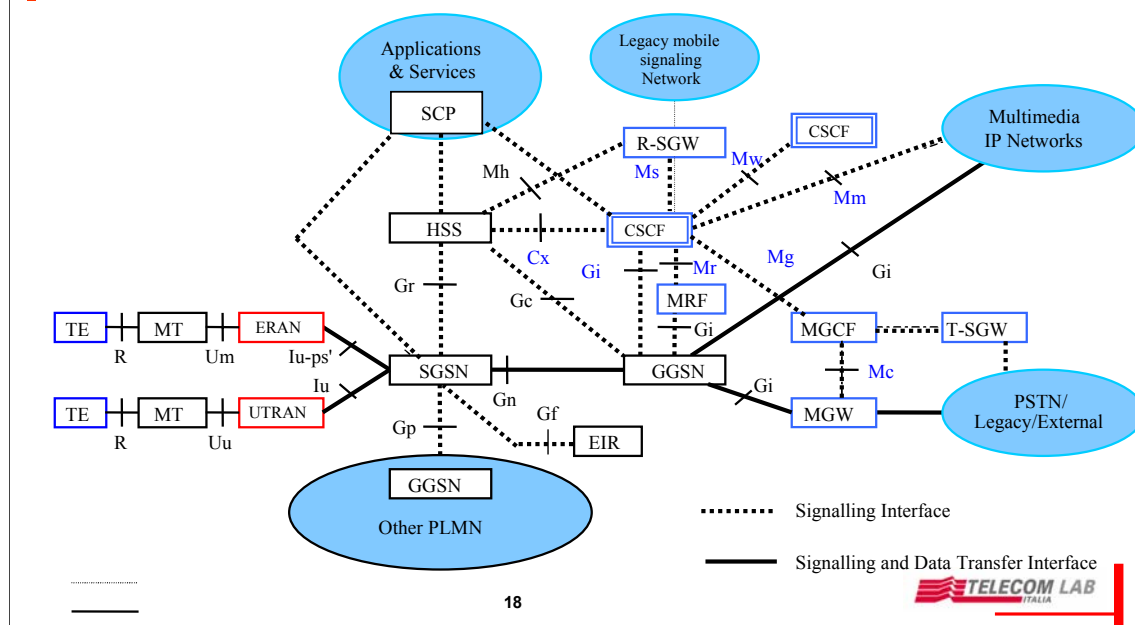
- Access independence
 - An operator should be able to offer services to subscribers regardless of how they obtain an IP connection (e.g. GPRS, fixed lines, LAN)
 - The IP MS should conform to IETF “Internet Standards”.
- IP multimedia applications shall not be standardised but it shall be possible to enable rapid service creation and deployment using service capabilities

In R5/R6 solutions, SGSN and GGSN supply the mobility management features and the activation of the PDP contexts as for the GPRS case.

Call control and transport functions are supplied by:

- the Call State Control Function (CSCF), able to deal with call routing, addressing functions, call set-up and call termination, billing support, interaction with HSS to receive user profiles
- the Transport and Roaming Signalling Gateways (R-SGW and T-SGW): the former acts as gateway for release 2000 users roaming in release 99 networks (mapping of SS7 on IP); the latter maps the PSTN/ISDN call associated signalling over the IP transport
- the Media Gateway Control Function (MGCF) which translates the messages received by T-SGW into IP Call Control messages for CSCF
- the Media Gateway (MGW) which terminates circuit connections over packet connections, controls all the needed transport entities (e.g. codec, echo cancellers).

The IP Multimedia domain

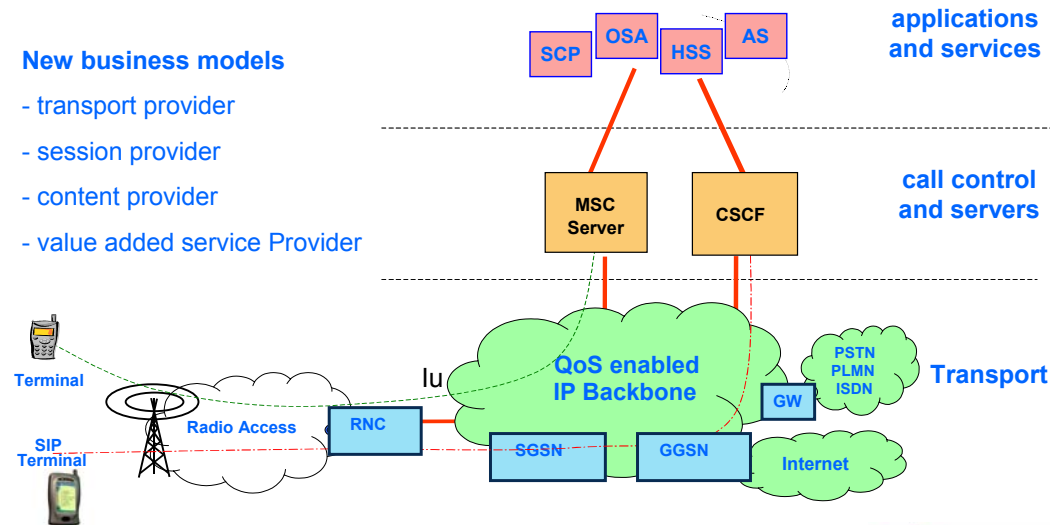


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Target UMTS solution: a new layered architecture



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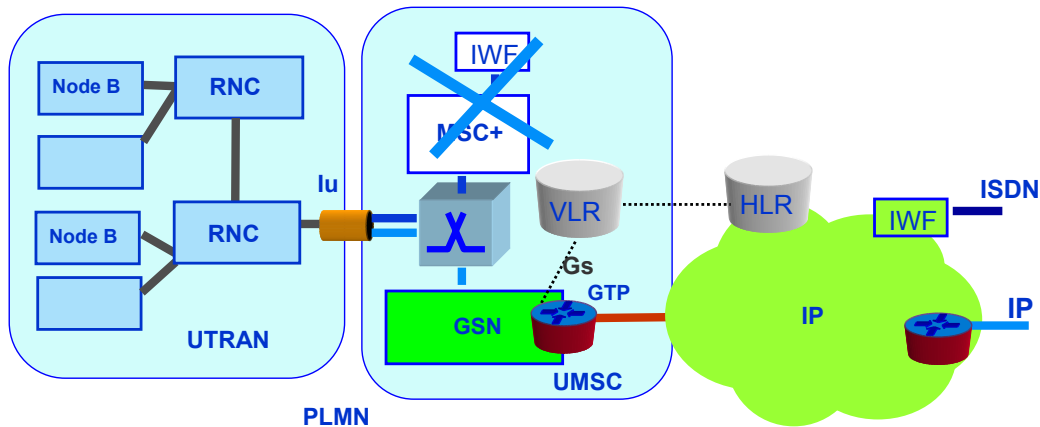
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The target UMTS solution is based on:

- Separation between Bearer level, Call control level and Service level:
 - the access to the IP Multimedia Subsystem (control) is supported by the PS domain (transport)
 - evolution of the CS domain with separation of transport and control
- Independence of services and technology
 - VHE/OSA and API for service creation
- Independence of access technology (RAN = UTRAN, GERAN).
- Separate functions that are likely to evolve independently
- End-to-end QoS support with high reliability and spectrum efficiency
- IPv6 is the basis of the ALL IP System: Huge growth of mobile Internet terminals would exhaust IPv4 address space. All wireless terminals will have WAP and GPRS. IPv6 brings enough IP addresses. Always-on connection establishes a variety of new services (Push, location-based, etc.).

Integration over IP backbone

.....the original IMS concept



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The target solution could fully use mobile IP protocols and some of the GPRS-based mechanisms can be abandoned (mobile IP is substituting GTP).

The tunnelling mechanisms and possibly the route optimisation schemes previously described are fully adopted to manage mobility up to GSN level. Within a given UMSC control context, mobility is managed through the continuous mobility control mechanisms, as described for the UMTS radio system.

A Home Agent is assigned to each operator and a Foreign Agent is associated to each UMSC control area.

Mobile IP (MIP) procedures are invoked whenever the mobile station is moving across different UMSC areas.

MIP procedures are used to manage the mobility between different UTRAN networks. GGSN and SGSN are playing the role of (substituted by) Home Agent and Foreign Agent entities respectively.

Voice over IP

- support offered by H323 (ITU) and SIP (IETF)
- H323 based on multimedia protocols (videoconference)
- SIP (Session Initiation Protocol) based on HTTP IETF; suited for creating, modifying and terminating sessions within a client/server model
- SIP identifies the called user and sets-up the communication through:
 - User Agent (controls the call on the client side)
 - Network Server, acting to update user location, to route SIP requests towards a new server, to direct requirements towards the user agent

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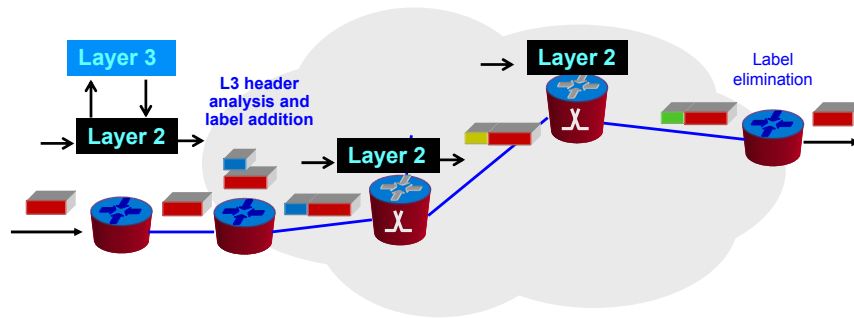
SIP can supply several services:

- user location services; whenever the SIP server receives a request, it localises the called party in its own domain by querying a registration server
- control of the call establishment and management
- Start of the multimedia session by making use of the Session Description Protocols (supplying media type, address, formats ports).

The service architecture is based on the Open Service Architecture concepts. It can be used not only to create services in a strict sense, but also to abstract some network features (e.g. call control, user location) with respect to the underlying network layers and features. This is done through Network Servers.

In SIP it is possible to create services by adding a service logic acting on the network server of the OSA.

The QoS problem on IP - MPLS



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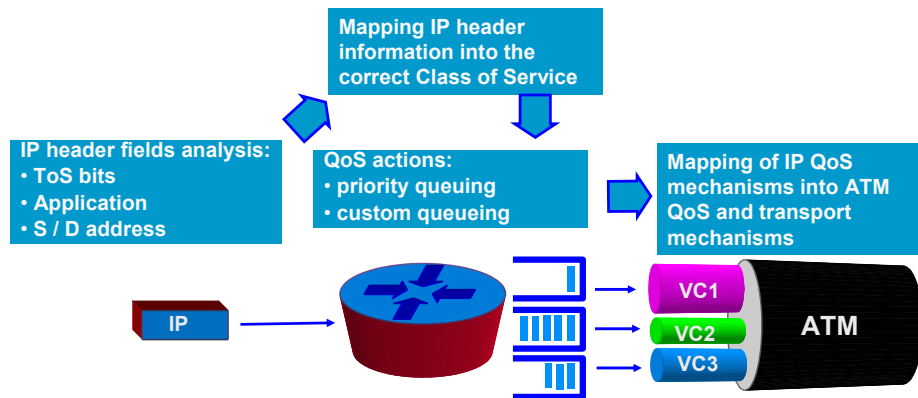
There are basic research interests having an impact on the *all IP* system development. They concern also the evolution of the IP-based mechanisms. Important issues in this context are:

- how limiting the additional IP headers on the air interface
- how satisfying the QoS requirements in an IP-based network; network edge control mechanisms, and network node control mechanisms are investigated (integrated service, differentiated service approaches respectively)
- how introducing routing schemes in the network (fixed and dynamic routing)
- how implementing Virtual Private Networks in an IP context through traffic segregation and tunnelling mechanisms.

The *Integrated Services* approach aims at implementing in an IP network, QoS levels equivalent to those typical of a circuit switched network. For the integrated service schemes, control is performed at the edges and an end-to end guarantee is constructed on the basis of service classes (see figure, where the QoS requirements are ensured through the underlying ATM Network).

As for differentiated service approaches, priority management resides in the single IP nodes within the network.

Improving QoS on IP: integrating IP and ATM mechanisms



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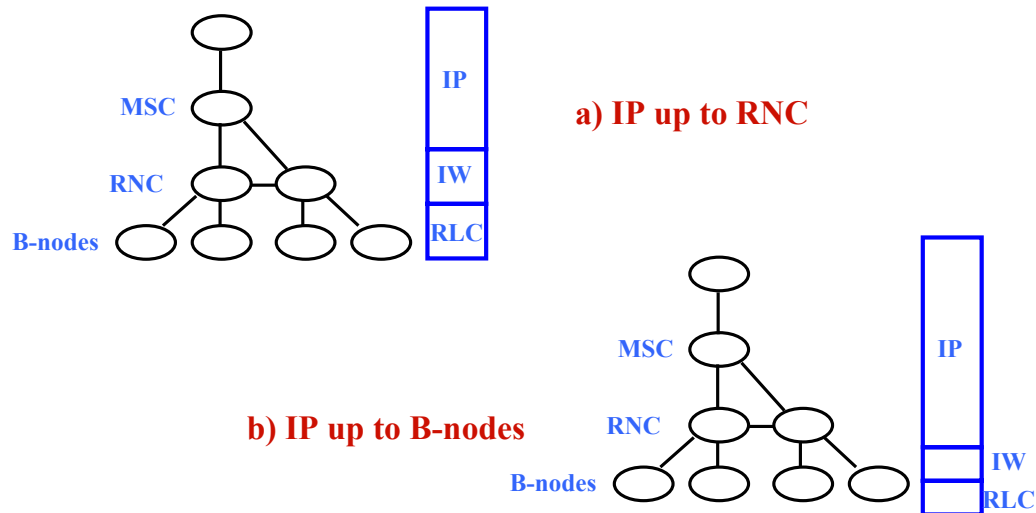
Benefits in terms of Quality of Services could be obtained by integrating IP QoS schemes and ATM QoS and transport mechanisms (e.g. through a Virtual Call-based bandwidth allocation, and the use of different transport classes ABR, UBR, CBR, VBR).

The first IP products integrating IP and ATM QoS mechanisms are now commercially available.

In the figure:

- ToS stands for Type of Service (information element of the IP header)
- S/D is the Source/Destination IP address.

Diffusion of IP networking



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The long term steps towards the IP penetration can have significant impacts on the network functions.

Extending the adoption of IP protocols below the IU interface, raises the problem of locating the inter-working functions with the radio protocols (radio link control). This location identifies the diffusion of the IP networking capability in the network.

In this choice a basic role is played by diversity mechanism; more precisely, the location of the IW functions depends on the alternative “closure” of the diversity mechanism in the up and down link:

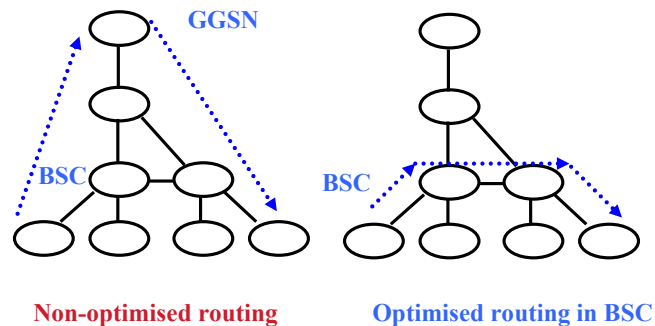
- if the diversity is entirely closed at B-node level, then the IP networking can include RNCs and the IW can be assigned to the B-nodes as well; this can be done if Layer 1 is terminated in B-nodes, where a distinction can then be done between voice and IP traffic
- if, conversely, the diversity is extended up to RNC, then the latter has to combine different macro-diversity branches and for delay and synchronisation constraints, the B-nodes-RNC transport cannot be assigned to IP mechanisms: IP must be stopped at RNC level and the IW functions are assigned to the same entity; this is the case when Layer 1 is terminated at RNC level.

IP networking to optimise routing for local traffic

Use of IP transport and networking



- optimal routing for local traffic
- how performing mobility functions in an IP-based solution



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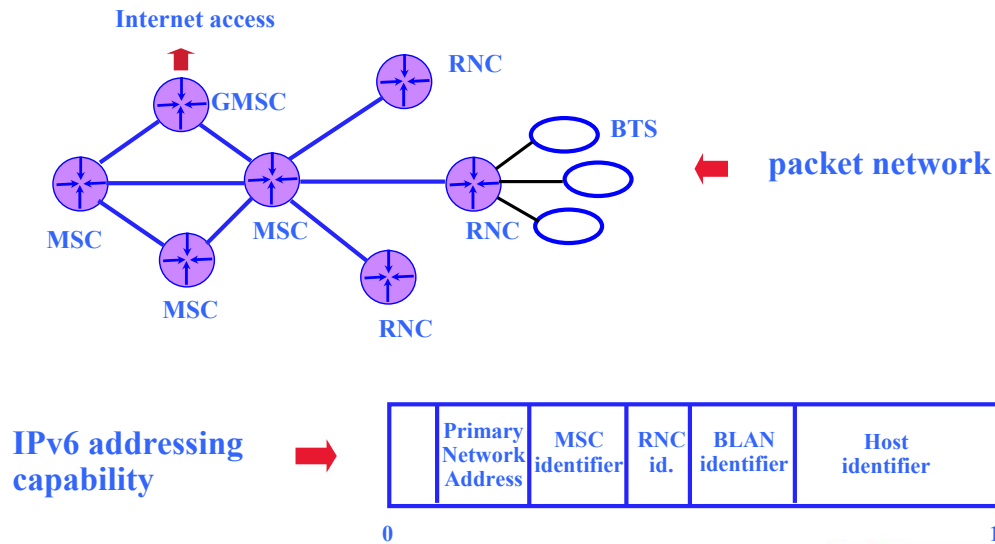
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As said before, in GPRS the networking features of IP are used only between GSN nodes and the routing to RNCs obeys rules of GSM (the packet is routed by following the mobile location mechanisms). The same holds for all those solutions confining the IP-based protocols above the Iu interface level.

Local routing is possible whenever IP networking and mobility mechanisms are available in RNC as well (figure).

However, the use of IP protocols raises at least the problem: how mobility control functions can be performed in a data network completely based on IP transport and networking and mobility features. This is another crucial element to be possibly investigated for the medium-long term evolutionary steps of UMTS.

Mobile IPv6 and third generation mobile



Some of the problems arising when a deeper penetration of IP in the mobile network is adopted are now considered. The assumption is then that both MSCs and RNCs implement IP routing functions to route packet traffic. (In addition, the necessary switching and control functions needed to deal with the circuit-switched traffic that are of course implemented in the same entities as far as the real time connection are carried in that way).

So, as for the packet traffic, third generation mobile system could be considered as an IP network based on AAL5+ATM techniques.

Routers are indicated in a simplified way as in the figure, reporting the name of the network node they refer to (MSC or RNC) just to highlight the hierarchical level where they are located.

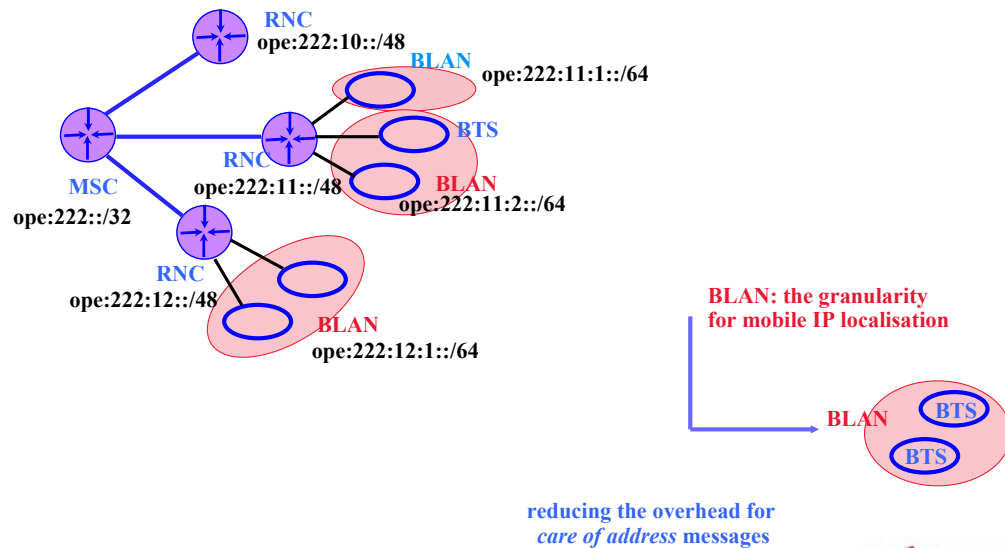
The GMSC represents the access point to Internet.

IPv6 addressing capability allows to assign each mobile user a static IP address and the network nodes can be assigned specific network addresses with no restriction.

The addressing can be organised in order to reflect the hierarchical order of the network architecture.

BLAN stands for BTS Local Area Network: it identifies the set of BTSs according to which the mobile terminal is localised according to mobile IP mechanisms.

Mobile IPv6 - addressing and localisation



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In the addressing scheme, ope identifies the operator network, the subsequent numbers, the MSC, the RNC and the BLAN.

Advantages of the mobile IP approach are:

- most of the MM functions that were located at application level are now moved at networking IP level (good for controlling the mobility at packet level)
- the approach is completely transparent with respect to the applications (mobility-independent applications).

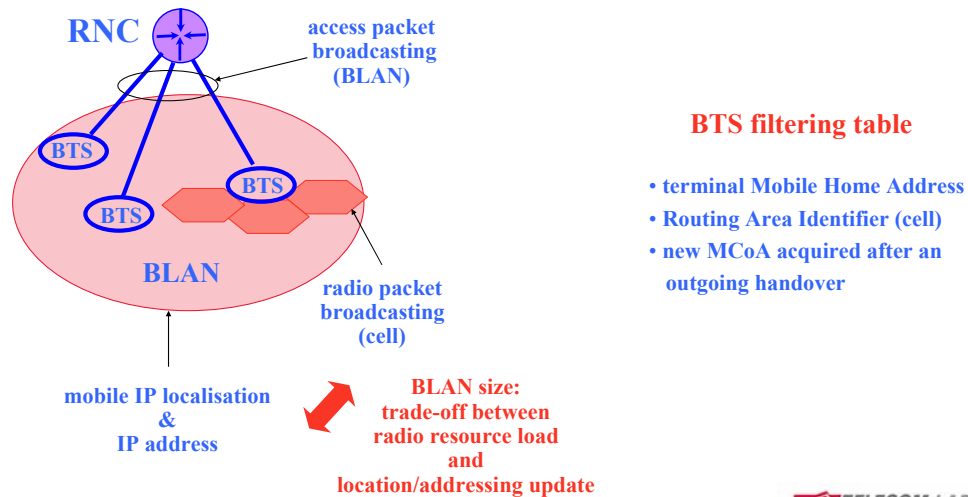
But it is necessary to avoid that Care of Address messages (to be exchanged in correspondence with the mobile movements) are overloading the network (in a mobile network we have to deal with a continuous mobility instead of a discrete mobility like in the fixed IP case).

In addition, the need to address the mobile terminal with the lowest possible granularity arises, since the single packet has to be multicast to the entire set of BTSs where the mobile terminal has been localised.

Of course the two requirements are in contradiction to each other, hence a compromise solution can be adopted:

- the location of the mobile terminal follows the granularity of a BTS Location Area Network (BLAN)
- the choice of the packets that are really to be sent on the air is left to the BTS that can recognise the cell where the mobile is (a BTS can control one or more cells).

Mobile IPv6 - balancing localisation load and radio resource use



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The above trade-off is played between two contrasting elements:

- the need of reducing the number of mobile-IP procedures to keep mobile location and address updated in function of the mobile movements
- the need of limiting the use of radio in the paging procedure.

It is reminded that a single diffusion mechanism is adopted in the circuit switched environment. In that case, it is natural to adopt location area of the same side of the paging area, because the paging process relates to a great amount of data to be exchanged along that connection; in addition, in that case, both paging and location updating are signalling procedures related to real (or potential) connections taking place outside the procedures themselves.

Conversely, in a packet context, the packet itself follows diffusion rules that are inherently similar to a paging event; this requires greater attention in limiting the packet diffusion area on the radio side.

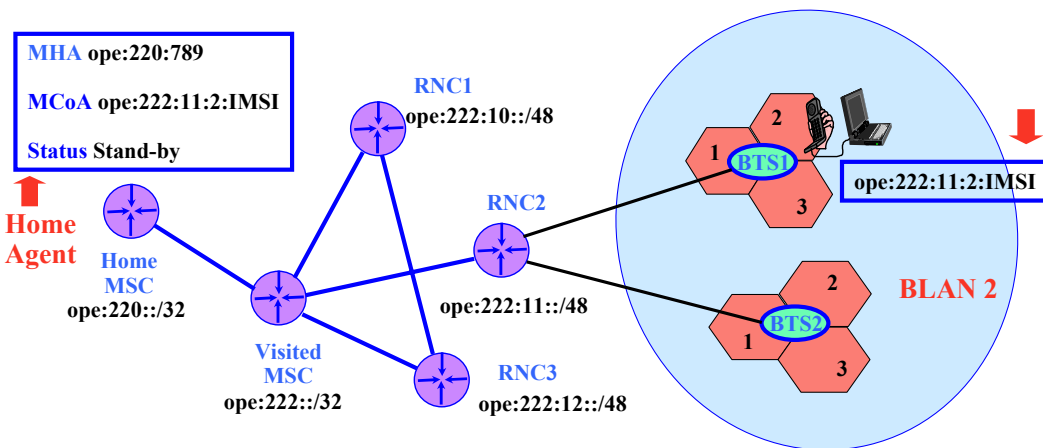
The solution proposed in the figure fulfils both requirements by identifying:

- a BTS Local Area Network (BLAN) sufficiently wide (to limit the location/addressing updating load)
- a packet diffusion area (Routing Area) sufficiently small, corresponding to a single cell (to limit the use of radio resources for packet transmission).

This choice induces some new functions in the BTS: this entity must perform a filtering role, limiting the packet transmission to the cell where the mobile terminal is roaming.

The filtering function is performed by keeping track of the mobile movements between cells (routing area) being the mobile identified with its Mobile Home Address (MHA).

Example of mobility procedure: attachment and stand-by state



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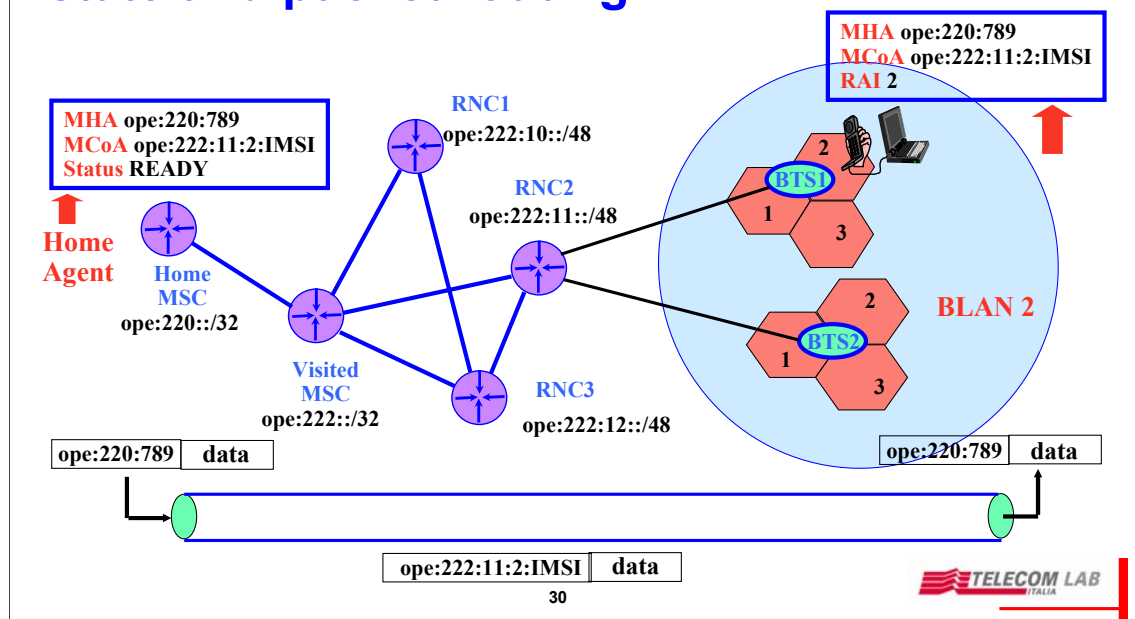
During the attachment, the mobile terminal associates its IMSI to the BLAN address and this is its Mobile Care of Address (MCoA).

The MSC updates its Binding Table by associating the Mobile Home Address to the above MCoA.

In this way the mobile is properly localised with the granularity of the BLAN it belongs to (the BLAN identifier is red by the mobile itself on the cell pilot channel).

In the Binding Table of the Home MSC, the status of the mobile terminal is maintained as IDLE: there is no need of identifying the mobile position with a greater precision since no packet has to be transmitted.

Example of mobility procedure: ready state and packet routing



When the terminal has to start a data transmission, it switches to the READY state.

From now on, the terminal location has to be identified with a greater precision (the routing area).

The READY status can be reached either through a paging procedure (incoming data traffic) or as a consequence of an independent access of the terminal to the network for data transmission.

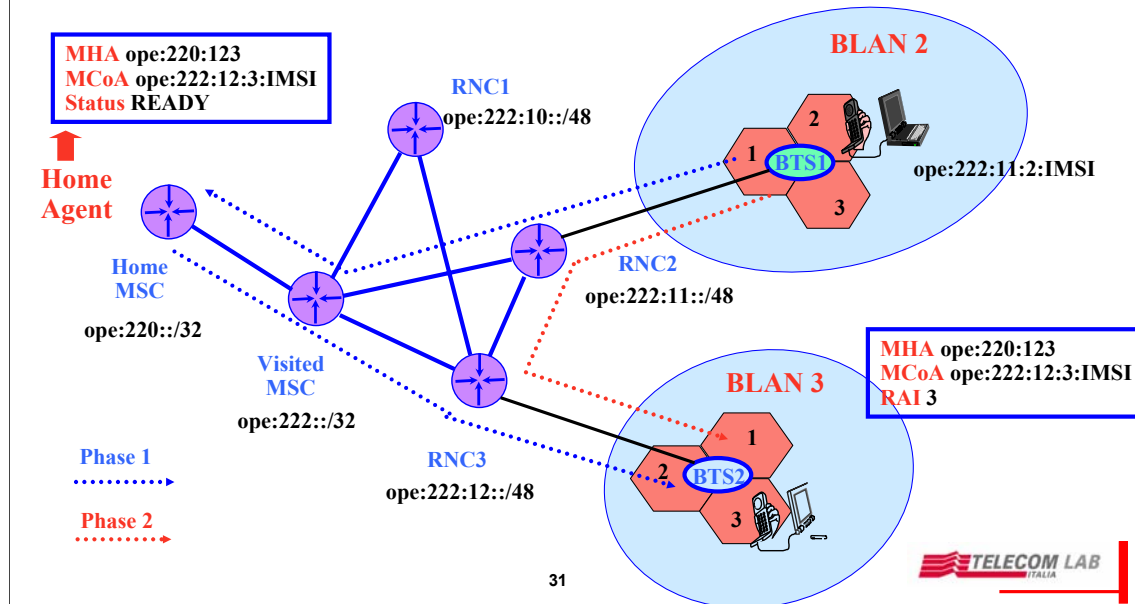
The READY state implies the opening of a filtering table in the serving BTS; the filtering table contains the MHA of the terminal: the BTS decides the packet transmission according to the presence of this address.

The incoming packet is tunneled by the Home MSC, acting as Home Agent, up to the mobile terminal; the tunnel is performed by using the MCoA, known to both the Home Agent and the terminal.

The packet maintains its IP Address.

The packet is diffused by RNC2 in all the BTSs belonging to BLAN2 (multicast) but only the BTS holding the MHA in its filtering table (BTS2 in the figure) will transmit the packet to the mobile terminal in the appropriate routing area..

Example of mobility procedure: optimising mobile-to mobile route

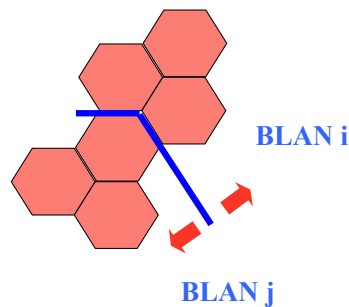


For mobile-to-mobile traffic, the first packet of the stream is sent, as usual, to the MHA; the latter tunnels the packet as in the previous case, by using as tunnelling address the MCoA of the requested mobile host (phase 1).

The requested mobile host recognises the reception of a *tunnelled* packet and then send directly to the calling host its own MCoA through a binding update message.

From that point on, the sending party fully utilises the IP networking to reach in an optimal way the destination host (phase 2).

Example of mobility procedure: handover and location updating



- Location updating: change of MCoA
- Handover:
 - updating of the filtering table (intra BTS)
 - creating a new entry for the filtering table (inter BTS)
 - as above +location updating (inter BLAN)

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It has been already highlighted how the mobile IP location is performed by following the BLAN granularity; the location updating procedure is started whenever a BLAN edge is crossed by the mobile terminal; it triggers a new registration and a change of the MCoA.

Conversely, the movements of the mobile terminal while a data transaction is in progress are done with a cell granularity; it can represent:

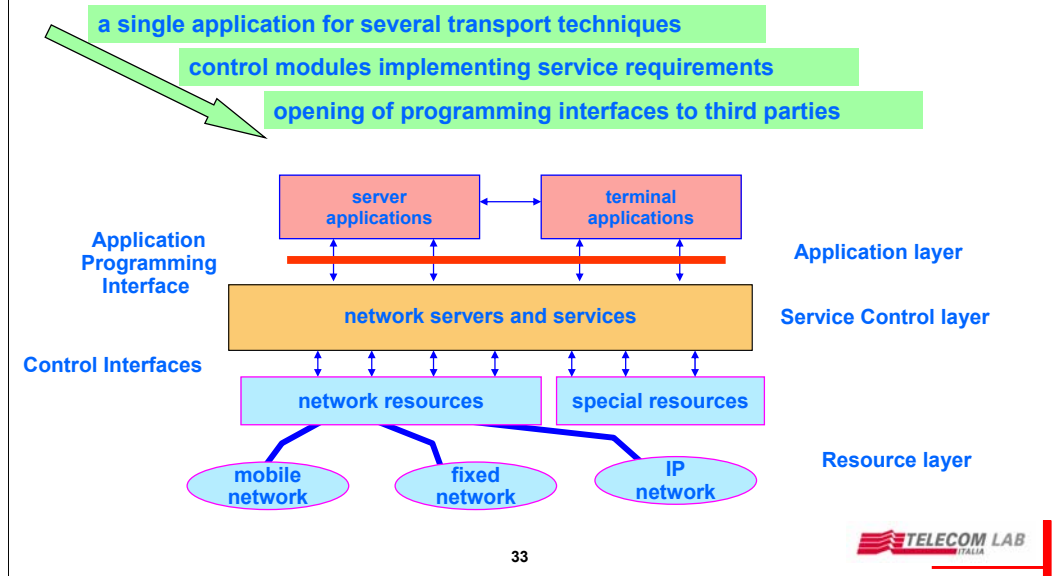
- a change of cell (routing area) inducing an updating of the Routing Area Identifier (RAI) in the filtering table of the BTS
- a change of BTS, inducing a deletion of the mobile-related entry in the old filtering table and the creation of a new entry in the filtering table of the visited BTS
- a change of BLAN, inducing the same filtering table updating and, in addition, an IP location updating with a corresponding change of MCoA.

During the handover execution, some packets addressing the old BTS, could be lost if they join the RNC when the packet transmission control has already been passed to the new BTS.

In order to avoid the packet loss, the old BTS could maintain for a given time after the handover execution, the new MCoA of the mobile terminal, in order to re-route the wrongly received packets.

The paging procedure can take place as for e.g., the GSM system (spreading of the paging message over the entire BLAN); it uses common signalling channels outside the data transmission; within the data transmission phase, the paging procedure can be carried through in-band signalling.

Service creation and control: the Open Service Architecture



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The *Open Service Architecture* co-exists with the *client-server* model, where the client and the server establish a functional relationship which is completely independent from the network functions and its configurations.

The introduction of such concepts is becoming in a gradual way, where first a *remote service execution* (client- server) concept and a *service emulation* concept are pursued (the latter is the reallocation of the service logic in the visited network).

OSA service models

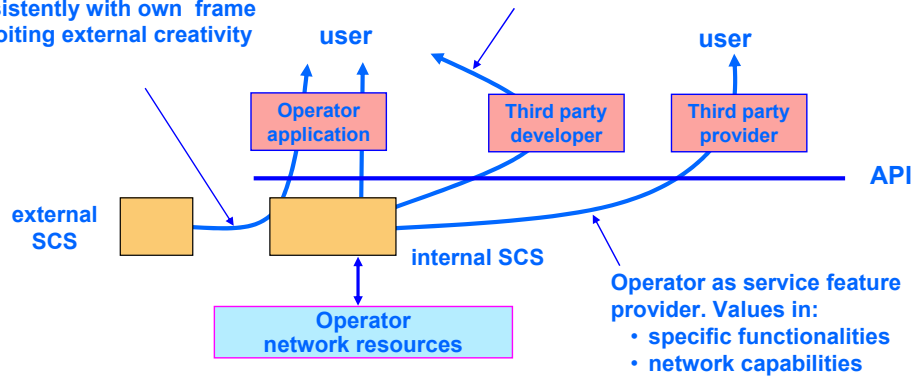
Operator as service broker.

Value in:

- enabling new applications consistently with own frame
- exploiting external creativity

Third party service developer. Value in:

- de-coupling from manufacturer
- external creativity



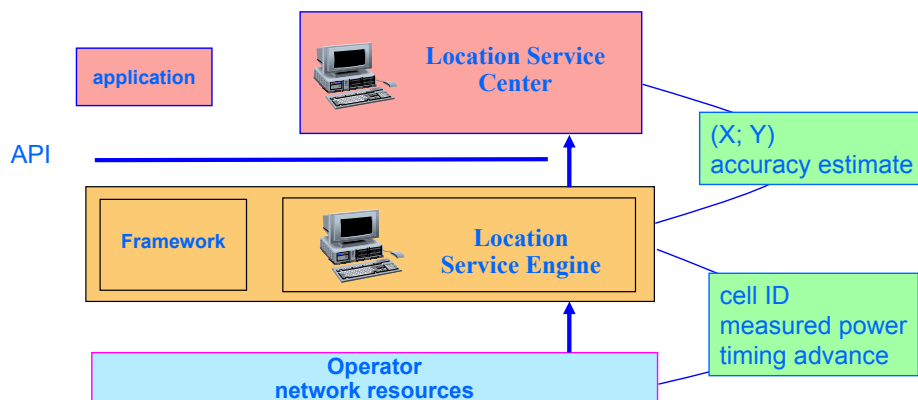
Operator as service feature provider. Values in:

- specific functionalities
- network capabilities

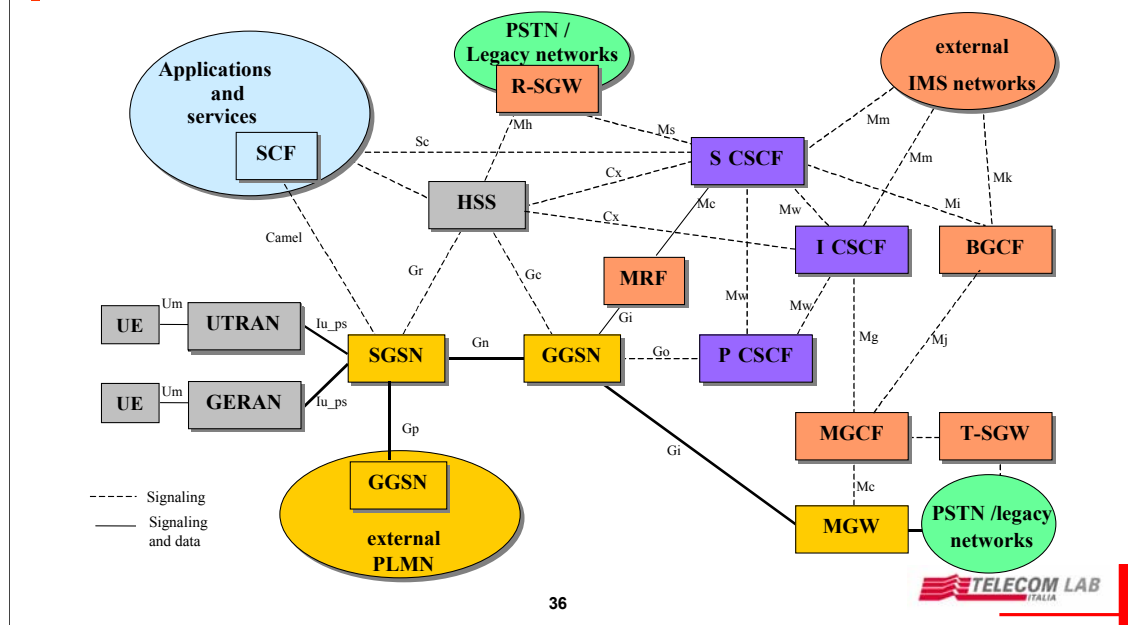
SCS: Service Capability Server

Localisation through Network data - an example of Service Capability Server

.....opening towards the application layer



IP Multimedia Subsystem (IMS)



The figure highlights the functions introduced in the IP Multimedia Service of UMTS. The heart of the IP multimedia domain architecture is the Call State Control Function (CSCF) which is a SIP server handling session set up and release, admission control, charging and security. In this way, the CSCF can be compared functionally to the MSC Server of the CS domain. The CSCF (Call Session Control Function) is operating on top of the basic IP connectivity services offered by the packet domain. It bases on the SIP protocol and provides mechanisms for advanced coordination of single/multiple session to support establishments of multimedia services for the end user.

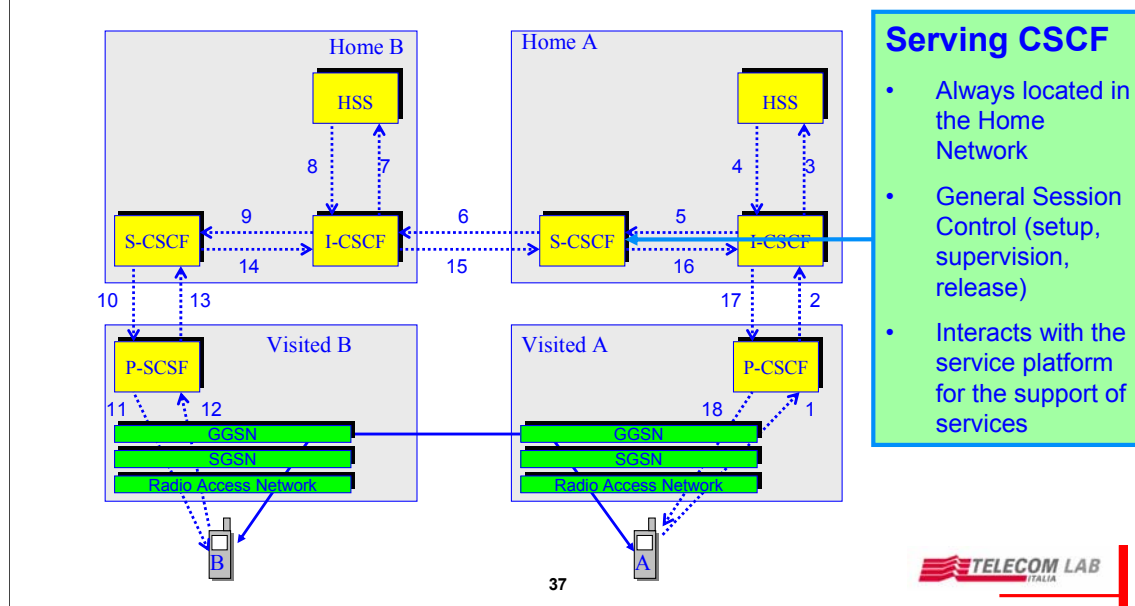
Logically, the CSCF can be divided into 3 components: Interrogating CSCF, Proxy CSCF and Serving CSCF that will be described later. The CSCF has an interface towards the Application and Service environment to provide value added IM services.

For sessions to/from legacy networks, the CSCF communicates with the Breakout Gateway Control Function (BGCF).

The Home Subscriber Server (HSS) is a master database storing subscription and location information for subscribers belonging to the network. The HSS is an upgrade of the HLR capable of handling SIP functions as well.

The GGSN node is directly connected to surrounding Multimedia IP networks on one side and can be connected to circuit switched networks via the Media Gateway. For traffic on the latter interface the Media Gateway will be controlled by a Media Gateway Control Function.

Different Kinds of CSCFs: S-CSCF



Currently, 3GPP has defined three different functional schemes which the CSCF will exhibit.

The Serving CSCF (S-CSCF), quite simply, provides users services.

The S-CSCF hosts the network functionality for the provision of calls/sessions. It interacts with the HSS and the application platform (not shown) to obtain subscriber data and for application service provision.

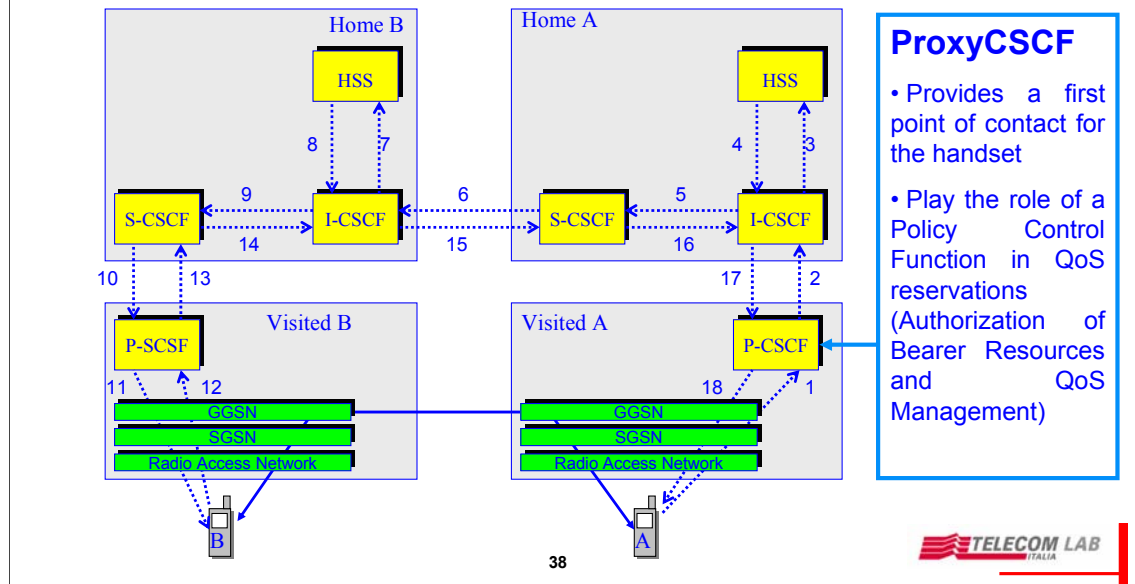
It holds registration state and call/session states.

It generates Call Detail Records (CDRs) for charging and resource utilization.

The S-CSCF will be useful in providing, for example: call forwarding when the terminal is not available, call barring, centralized speed dial lists, VPN services, etc.

The architecture is based on the principle that the service control for Home subscribed services for a roaming subscriber is in the Home network, e.g., the Serving-CSCF is located in the Home network.

Different Kinds of CSCFs: P-CSCF



The Proxy CSCF (P-CSCF) provides a first point of contact for the handset. All signalling to and from the handset goes through the P-CSCF.

Its address is discovered by UEs following PDP context activation, using the mechanism “CSCF Discovery”. The P-CSCF behaves like a Proxy, i.e. it accepts requests and services them internally or forwards them on, possibly after translation. The P-CSCF may also, in abnormal conditions, terminate and independently generate SIP transactions.

It generates Charging Data Records.

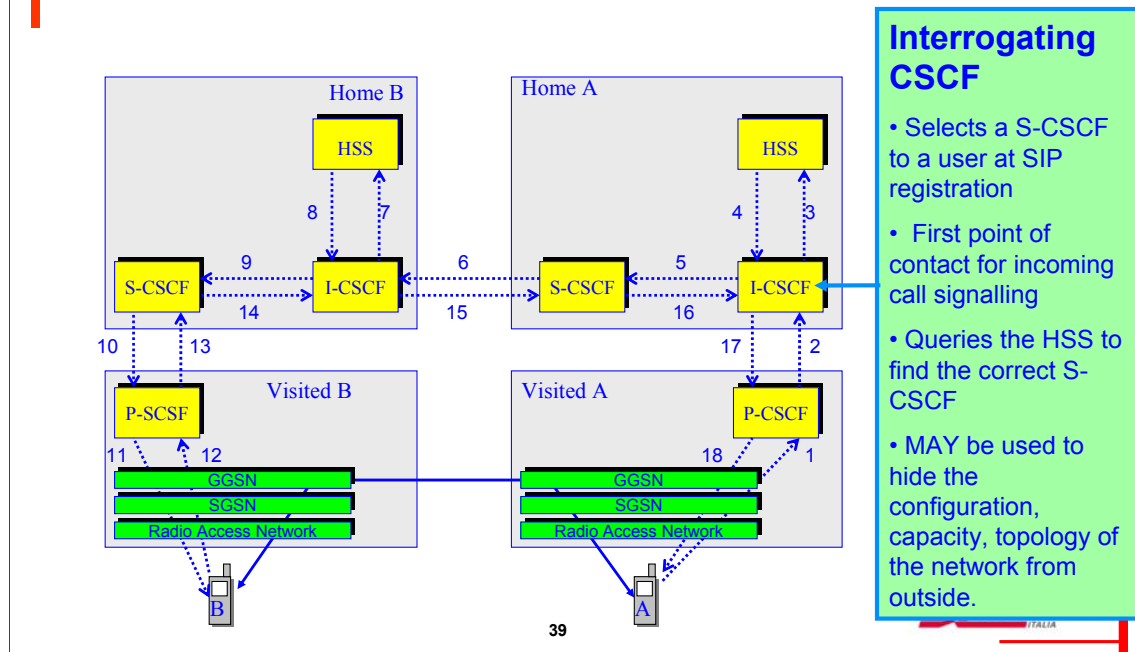
The Policy Control Function (PCF) is a logical entity of the P-CSCF that plays a role in quality of service reservations (authorization of bearer resources and QoS Management) to enable coordination between events in the application layer and resource management in the IP bearer layer.

The PCF makes decisions in regard to network based IP policy using policy rules, and communicates these decisions to the GGSN, which is the IP Policy Enforcement Point (PEP). The interface between the PCF and GGSN is specified within 3GPP, and named Go interface: it supports the transfer of information and policy decisions between the policy decision point and the GGSN.

The PCF makes policy decisions based on information obtained from the P-CSCF. In the P-CSCF(PCF), the application level parameters (e.g. Session Descriptor Protocol information) are mapped into IP QoS parameters.

The P-CSCF (PCF) must be in the same domain as the GGSN.

Different Kinds of CSCFs: I-CSCF



The Interrogating-CSCF (I-CSCF) is the contact point within an operator's network for all connections destined to a subscriber of that network operator, or a roaming subscriber currently located within that network operator's service area.

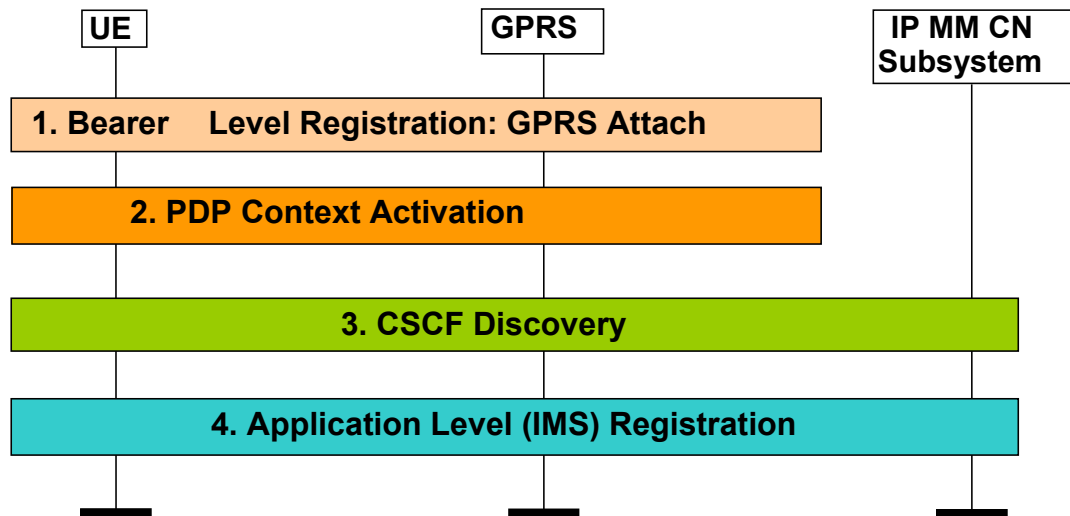
It is mostly a load distribution node. In conjunction with the HSS, allocates subscription information onto appropriate Serving CSCFs. The HSS keeps track of this information so that all transactions and all calls for the same user go through the same service node.

Since DNS allows simple statistical distribution among identical nodes, distributing load among the I-CSCFs is quite simple. But if all connections destined to a subscriber relied on statistical distribution, it wouldn't be possible to allocate subscriptions on appropriate serving nodes according to their capabilities, nor to be able to keep call state information between transactions.

The I-CSCF generates CDRs.

The operator may use the I-CSCF or other techniques to hide the configuration, capacity, and topology of the network from the outside. When the I-CSCF is chosen to meet the hiding requirement then for sessions traversing across different operators domains, the I-CSCF may forward the SIP request or response to another I-CSCF allowing the operators to maintain configuration independence.

IMS Registration



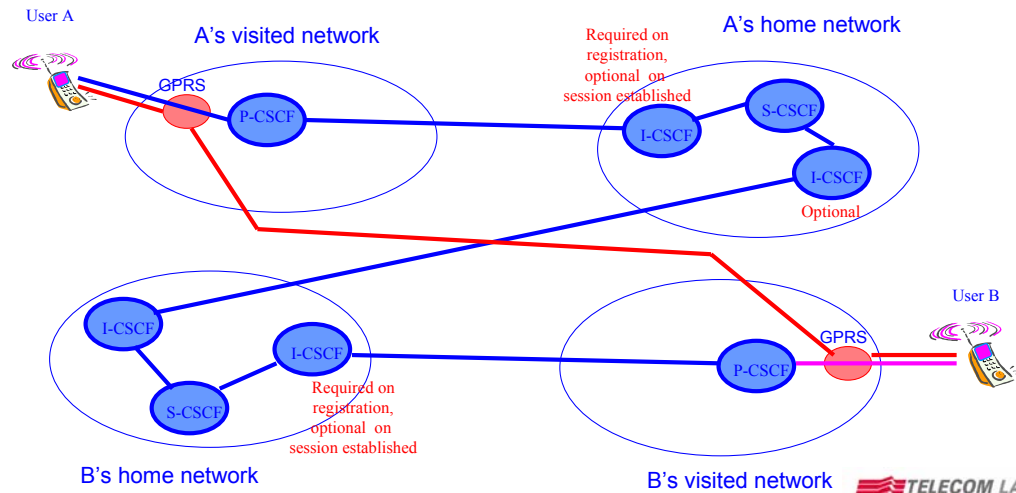
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In order to access IMS services, the UE performs the following logical steps:

1. bearer level registration: upon switch-on or upon explicit indication from the user, the terminal performs a GPRS Attach
2. PDP context Activation: this step is needed to acquire an IP address and to establish IP connectivity to exchange the necessary SIP signalling
3. Proxy CSCF discovery procedure: it allows the UE to find out the first contact point to send initial registration requests to. The Proxy-CSCF discovery shall be performed after GPRS Attach using one of the following mechanisms: A) DHCP, to provide the UE with the domain name of a Proxy-CSCF and the address of a Domain Name Server (DNS) that is capable of resolving the Proxy-CSCF name; B) transfer a Proxy-CSCF address within the PDP Context Activation signalling to the UE.
4. UE performs application level registration by providing the IP address and the Home Domain Name to the Proxy CSCF discovered at step 3; the Proxy CSCF forwards the registration request to an I-CSCF of the home network selected on the base of the Home Domain Name. When the I-CSCF receives the registration information, it selects a suitable S-CSCF that will complete the registration procedure. As a result of the successful registration the S-CSCF will store Subscriber information and Proxy address/name, the P-CSCF will store the home Network Entry point and UE Address, and the UE will store the Proxy Name/Address.

Example of allocation of CSCFs (mobile to mobile case)



All blue lines represent SIP signalling path.

All red lines represent the bearer streams.

P-CSCF and S-CSCF are bound at registration time. Therefore a UE will use the same P-CSCF and S-CSCF until a new registration occurs. I-CSCF is only retained in this binding if the home network wishes to hide its S-CSCF configuration.

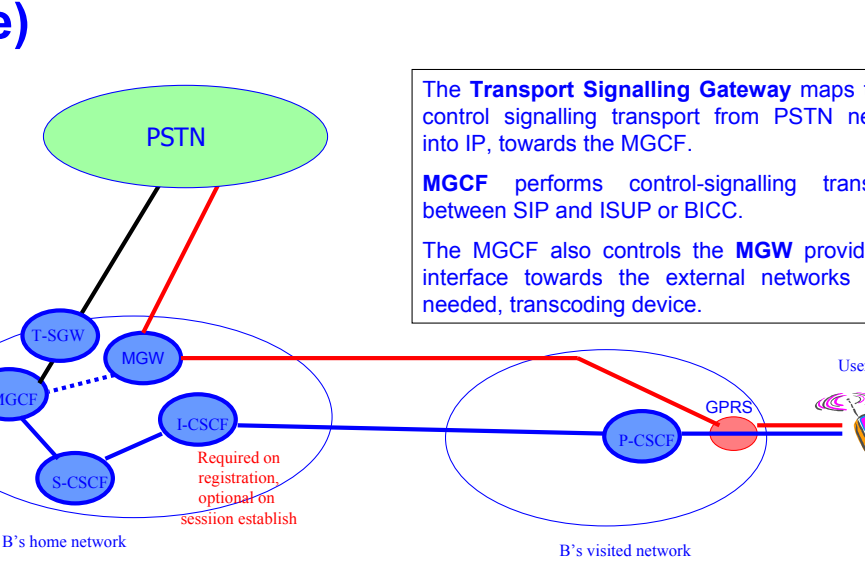
Once the S-CSCF is reached in establishing this session, it must find the remote S-CSCF handling the user B. It does so by sending the INVITE request to the remote home network, where an I-CSCF performs the necessary lookups to locate the S-CSCF of the remote user. This I-CSCF may also perform hiding.

The only routing that is not fixed at registration time is the S-CSCF to S-CSCF portion.

All session signalling is transferred via the home networks of each subscriber.

The bearer stream may be carried directly between the current network of each subscriber or via the respective home networks in case the PDP contexts carrying bearer streams terminate in the home GGSNs.

Example allocation of FEs (PSTN to mobile case)



The diagram illustrates the functional architecture for a PSTN to mobile call. It shows two main network domains: 'B's home network' and 'B's visited network'.

B's home network contains the following components:

- PSTN** (Public Switched Telephone Network) represented by a green oval.
- T-SGW** (Transport Signalling Gateway) represented by a blue circle.
- MGCF** (Media Gateway Control Function) represented by a blue circle.
- MGW** (Media Gateway) represented by a blue circle.
- I-CSCF** (Interrogating Call Session Control Function) represented by a blue circle.
- S-CSCF** (Serving Call Session Control Function) represented by a blue circle.

B's visited network contains the following components:

- P-CSCF** (Proxy Call Session Control Function) represented by a blue circle.
- GPRS** (General Packet Radio Service) represented by a red circle.

User B is represented by a mobile phone icon connected to the GPRS interface.

Connections and Signaling:

- A black line connects the **PSTN** to the **T-SGW**.
- A red line connects the **PSTN** to the **MGW**.
- A red line connects the **MGW** to the **P-CSCF** in the visited network.
- A blue line connects the **P-CSCF** to the **I-CSCF** in the home network.
- A blue line connects the **I-CSCF** to the **S-CSCF**.
- A blue line connects the **S-CSCF** to the **MGCF**.
- A dotted blue line connects the **MGCF** to the **MGW**.

Text Box:

The **Transport Signalling Gateway** maps the call control signalling transport from PSTN networks into IP, towards the MGCF.

MGCF performs control-signalling translations between SIP and ISUP or BICC.

The MGCF also controls the **MGW** providing the interface towards the external networks and, if needed, transcoding device.

Legend:

- Required on registration
- optional on session establish

Page Information:

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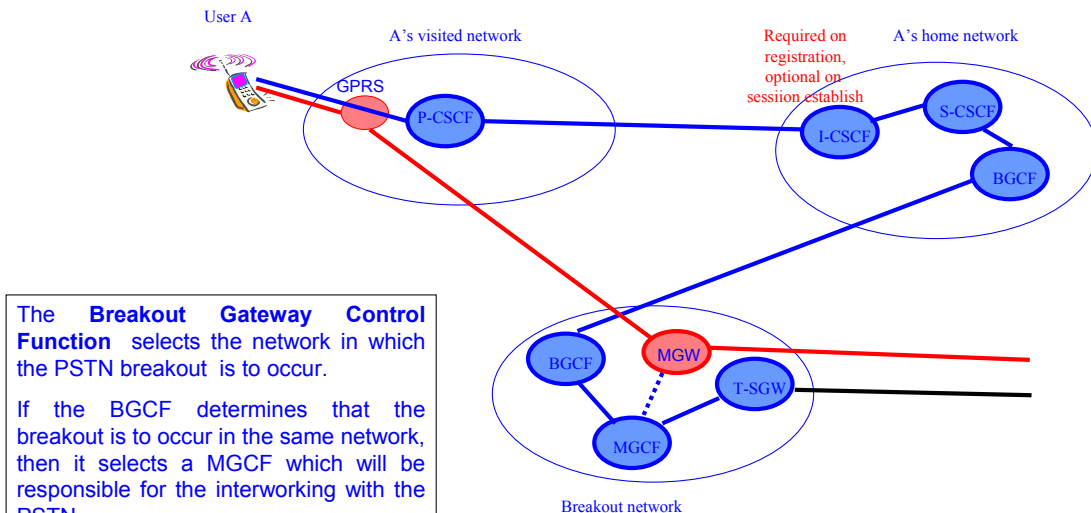
All red lines represent the bearer streams.

MGCF - Media Gateway Control Function - provides the signalling inter-working with the PSTN, or with the circuit switched side of 3GPP. This controls, via H.248/MEGACO, the MGW - Media Gateway - which provides the bearer stream inter-working.

PSTN and CS inter-working is with ISUP or BICC.

Pag. 42

Example allocation of functional entities - mobile to PSTN



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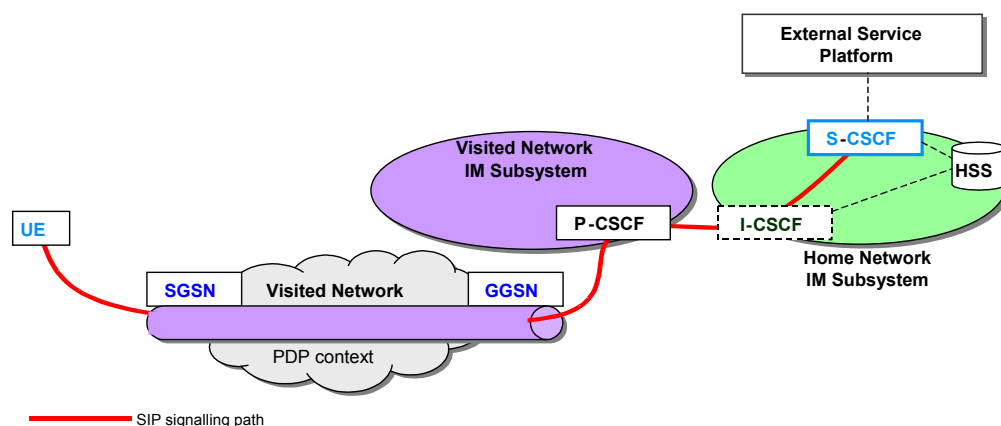
All blue lines represent usage of SIP signalling.

All red lines represent the bearer streams.

In addition, we also have the following function for access to the PSTN and the circuit-switched network:

BGCF - Breakout Gateway Control Function - selects the network, or MGCF, in which PSTN breakout is to occur. If breakout occurs in the local network, then only one BGCF is present in the path; if the breakout occurs in another network, then multiple BGCFs will be provided. This is determined by the roaming agreements between the network operators.

Home Control of Service



The Serving-CSCF is located in the Home network

The architecture is based on the principle that the service control for Home subscribed services for a roaming subscriber is in the Home network.

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The architecture is based on the principle that the service control for Home subscribed services for a roaming subscriber is in the Home network, e.g., the Serving-CSCF is located in the Home network. This architectural choice is also referred to as “home control of service”.

The reasons behind the Home Control choice are:

- to ensure end users experience to be consistent (unaware of change in domains-within or outside of home operator’s environment)
- minimize dependencies between Home and Visiting networks regarding call/session control and services triggering
- allow support for Operator differentiation.

At the beginning of IMS standardization process, 3GPP had focused on two possible Service control scenarios for IM CN subsystem, Home or Visited network control of sessions; then, the decision to select one of the models would have been done by the Home operator during IM registration based on some specific criteria.

Eventually, the majority of 3GPP companies concluded that there was no special gain foreseen from the Visited control model, which could not be achieved from the Home control model and proposed 3GPP to remove the Visited control option from IM CN subsystem.

Then, Home control of services is the only option supported within 3GPP Rel 5 and it’s likely to stay the only option also for subsequent releases.

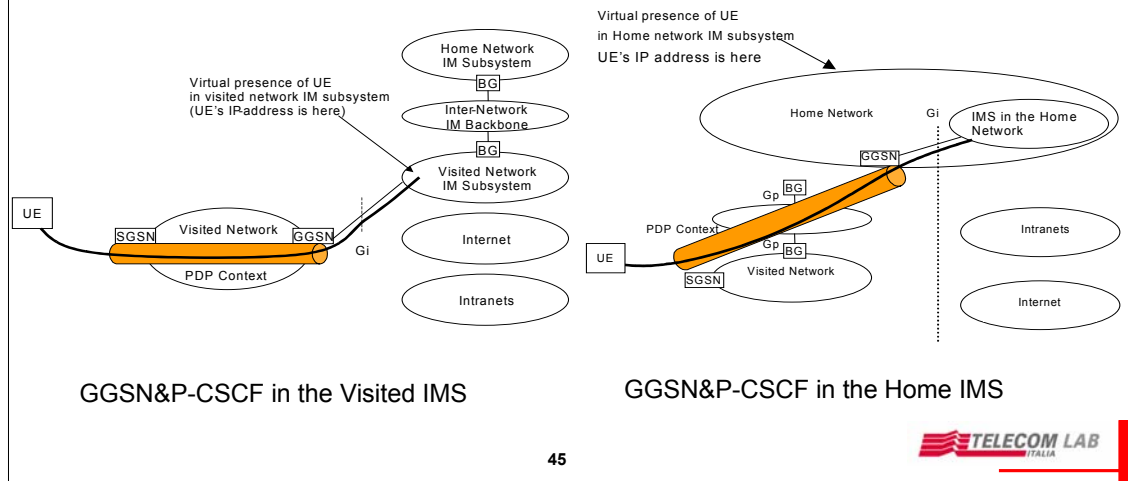
A Proxy-CSCF is supported in both roaming and non-roaming case, even when the Serving-CSCF is located in the same IM CN SS.

The use of additional CSCFs, that is Interrogating-CSCFs, to be included in the SIP signalling path is optional. Such additional CSCFs may be used to shield the internal structure of a network from other networks.

IP Address and routing

A UE accessing IM-Subsystem services requires an IP address, that is obtained using an appropriate PDP-context.

It is possible to connect to a GGSN either in the VPLMN or the HPLMN.



As specified by the GPRS procedures, the UE shall acquire the necessary IP address as part of the PDP context activation procedure, which includes, or is immediately followed by, the P-CSCF discovery procedure.

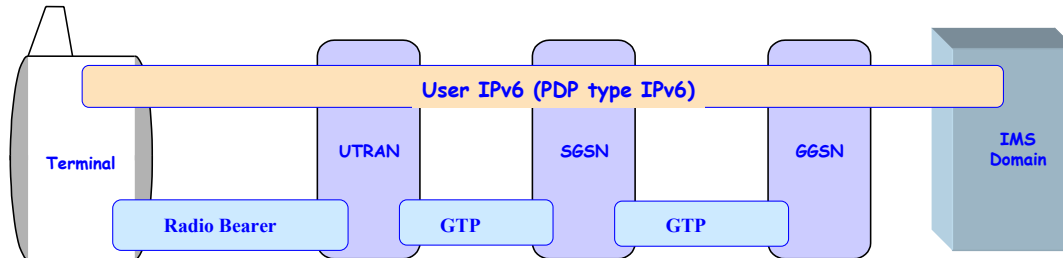
Example of possible connections between the UE and the IM Subsystem are shown:

- UE Accessing IM Subsystem Services with GGSN in the visited network
- UE Accessing IM Subsystem Services with GGSN in the home network

For routing efficiency the PDP context may benefit from being connected through a GGSN in the visited network. However, having a GGSN in the Home network is in line with the present GPRS solution and may enable network (i.e. RAN + SGSN) sharing.

Since the Policy Control Function (PCF) within the Proxy-CSCF applies policy to the bearer usage in the GGSN, the P-CSCF (PCF) is in the same domain as the GGSN or has a trust relationship with the GGSN. However, currently in IETF, inter-domain policy interactions are not defined.

IPv6 - Transport of user IP packets in UMTS



IP packets to/from the terminal are tunneled through the UMTS network, they are not routed directly at IP level

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The UMTS/GSM architecture shall support IPv4 / IPv6 based on the statements below:

- IP transport between network elements of the IP Connectivity services (between RNC, SGSN and GGSN) and IP transport for the CS Domain → both IPv4 / IPv6 are options for IP Connectivity
- IM CN subsystem elements (UE to CSCF and the other elements e.g. MRF):
 - The architecture shall make optimum use of IPv6.
 - The IM CN subsystem shall exclusively support IPv6.
 - The UE shall exclusively support IPv6 for the connection to services provided by the IM CN subsystem.

Note: the exact set of the functionality available in the whole Ipv6 protocol suite (such as IPSec, IP multicast etc.) that will be mandatory in R5 is For Further Study.

- Access to existing data services (Intranet, Internet,...) → the UE can access IPv4 and IPv6 based services.

A dedicated design team within the IETF IPng working group has produced an Internet-draft formulating some recommendations to 3GPP about the use of IPv6 in GPRS.

Mobility

The IP multimedia subsystem utilizes the PS domain to transport multimedia signalling and bearer traffic.

The PS domain maintains the service while the terminal moves and hides these moves from the IP multimedia subsystem

There are two levels of mobility in the PS domain (UMTS): Micro-mobility, where the UE location changes are within RA/LA (Registration Area/Location Area); Macro-mobility that takes place between RA/LAs. In both cases...

...The Proxy-CSCF, behind the GGSN, acts as an anchor point.

Hence there is no need for the CSCF to be aware of the changes in RA/LA. Moreover, SIP is an application level protocol and SIP must not be aware of the link layer or network layer mobility issues implied by changes in RA and LA.

As specified by the GPRS procedures, the UE shall acquire the necessary IP address(es) as part of the PDP context activation procedure(s).

Identification of users

Every IMS subscriber has a Private User Identity (IMPI).

- The IMPI is assigned by the home network operator, and used for e.g. Registration, Authorisation, Administration, and Accounting purposes
es. *user_name* or *user_name@operator*

Every IMS user has one or more Public User Identities (IMPU).

- IMPUs are used by any user for requesting communications to other users. E.g. this might be included on a business card.
- Both Telecom or Internet numbering scheme are supported.
 - » E.164: e.g. + 39 011 228 5555
 - » SIP URL: e.g. sip: mariagiovanna.pautasso@tilab.com
- At least one Public User Identity shall be securely stored on the UICC (it shall not be possible for the UE to modify the Public User Identity)

“The home network operator is responsible for the assignment of the private user identifier, and public user identifiers.”

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Every IM CN subsystem subscriber shall have a private user identity (IMPI).

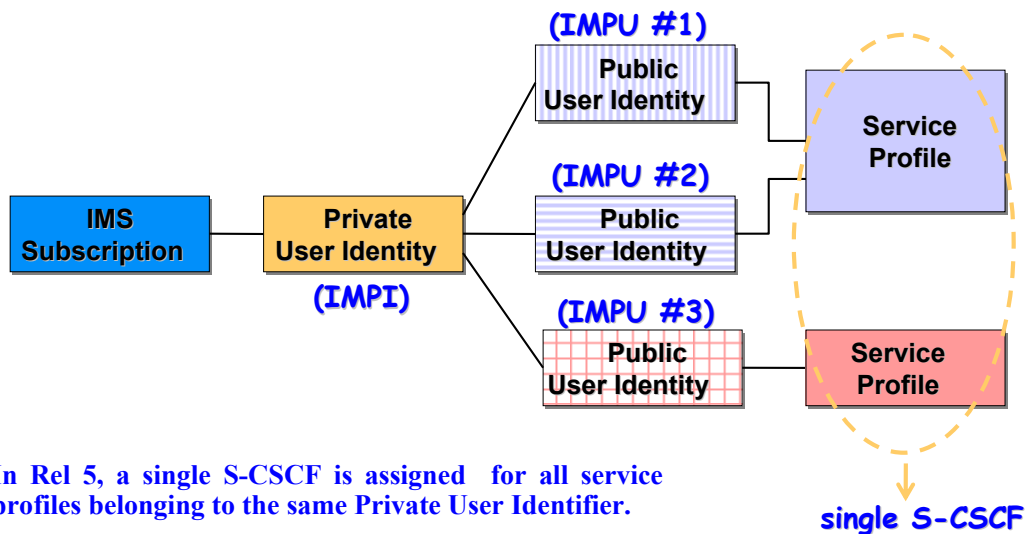
The IMPI shall be contained in all Registration requests passed from the UE to the home network. The IMPI is a unique global identity defined by the Home Network Operator, which may be used within the home network to uniquely identify the user from a network perspective. The IMPI shall be permanently allocated to a user (it is not a dynamic identity), and is valid for the duration of the user's subscription with the home network. The IMPI is authenticated only during registration of the subscriber (including re-registration and de-registration).

Every IM CN subsystem subscriber shall have one or more public user identities.

At least one Public User Identity shall be securely stored on the USIM (it shall not be possible for the UE to modify the Public User Identity), but it is not required that all additional Public User Identities be stored on the USIM. It shall be possible to register globally (i.e. through one single UE request) a subscriber that has more than one public identity via a mechanism within the IP multimedia CN subsystem. This shall not preclude the user from registering individually some of his public identities if needed. Public User Identities are not authenticated by the network during registration. The S-CSCF shall support the ability to translate the E.164 address contained in the “tel:” format to a SIP URL, using DNS translation mechanism. If this translation fails, then the session may be routed to the PSTN or appropriate notification shall be sent to the mobile.

The home domain name of the subscriber shall be stored securely on the USIM, (it shall not be possible for the UE to modify the home domain name).

Relations between IMPI and IMPU



In Rel 5, a single S-CSCF is assigned for all service profiles belonging to the same Private User Identifier. Assignment of multiple S-CSCFs should not be precluded in future releases of the IMS

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Subscribers may have different service profiles. Each Public ID belongs to a single service profile, but a single service profile may have several Public IDs. All public user identities that are associated with the same profile should have the same set of services.

The following example should help clarify the above concepts.

As a user of the existing wireless services and the future IMS services, I could have the following public user IDs:

- +39.123.456.789 (My existing E.164 number for my business mobile phone)
- mariagiovanna.pautasso@tilab.com (My existing business email/URL address)
- marpy@somenetwork.net (A possible personal future email/URL address)

For calls initiated from the IMS compatible mobile device, I would indicate at call origination if this was a business call or a personal call so that the charges associated with this call could be applied to the correct account. Incoming business calls to my mobile device could be directed to either my business E.164 number (e.g. +39.123.456.789) or to my business URL address (e.g. mariagiovanna.pautasso@tilab.com). Incoming calls addressed to either of these public IDs would receive the same exact call handling as both of these public IDs are alias identifications to my user profile for business calls.

Incoming personal calls to my mobile device would be directed to my personal URL address (e.g. marpy@somenetwork.net). Since this incoming call is addressed to a different address then the business calls, these incoming personal calls could be distinguished from business calls. Consequently, a completely separate user profile for personal calls could be applied. Based upon these separate user profiles the following capabilities would be available: e.g Business and personal calls charged to separate accounts, Different call forwarding handling, Business services (e.g., VPN) only associated with business account.

My business and personal profiles would be activated and deactivation independently. The activation and deactivation of my user profiles would be based upon my input from the mobile device (e.g. soft-keys) or based upon time of day based service. A default user profile activation, which I would be able to configure, could be associated with the mobile power on event.

IMS SIM (ISIM)

In the PS domain, the service is not provided until a security association is established between the mobile equipment and the network.

IM CN subsystem is essentially an overlay to the PS-Domain and is not embedded in the SGSN or GGSN nodes consequently a second security association is required between the multimedia client and IM CN subsystem before access is granted to multimedia services.

SA3 has defined a security architecture that includes the addition of a new logical entity on the UMTS UICC. This new entity is called “**ISIM**” and contains parameters and keys specific to the IMS:

e.g. (at least) IMPI; Home Network Domain Name; Algorithms and Authentication Key

“It is necessary for this subscription information to be independent of the corresponding USIM data to support:

-access network independence

-UE split.”

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3GPP “Security aspect group” introduced the concept of a logically independent ISIM to represent the IMS subscription and its associated data. In principle, it could be necessary for this subscription information to be independent of the corresponding USIM data, to support access network independence and UE split.

Private and public identifiers for the IP multimedia Subsystem should be securely stored in the USIM. But according to the above view, private and public identifiers for the IP Multimedia Subsystem are independent of the USIM and should be stored in the ISIM instead of USIM. The ISIM would also store an authentication key K for IMS use, and should store other data related to authentication and privacy. Also IMS subscription related data should be stored in the ISIM.

Even though it is likely for Release 5 that the ISIM and USIM will be implemented on the same UICC, and probably provisioned by the same provider, separating these functions at a future date would be very difficult if this logical separation is not in place from the start. However, such separation may not be convenient for existing mobile network operators:

- One of the successes of GPRS has been that it is possible to use GPRS mobiles with unmodified SIM cards. It seems sensible to ensure that something similar can be done for IMS, namely that IMS mobiles can work with an unmodified R'99 USIM.
- The proponents of “access independence” to IMS do not want to force every IMS terminal to have a USIM. The access technology might, for example, be Wireless LAN. Although there does not appear to be any technical problem with mandating the use of a USIM, there may be minor operational issues (e.g. would every IMS-provider be able to get a set of IMSI from which to derive IMPI?).