Technical Specification Group Services and System Aspects **_TSGS#12(01)0318_**
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Source: AT&T Wireless (ileana.leuca@attws.com)
Title: IETF-3GPP Report
Document for: Discussion

This document highlights the main areas of collaboration, the results of the last three months and also some recommendations derived from this experience.

Summary:

- 3GPP participated in the last IETF meeting - in March - with two presentations supported by Keith Drage, covering the IMS architecture and Thadeus Kobylarz covering the charging requirements.
  Note: it was very useful to present the 3GPP IMS architecture and to show how, using the signaling path, many networks/operators could potentially be involved in a call. It was agreed that several issues need settlement (e.g. notification from a UA in a proxy-initiated de-registration may use the SIP presence draft as a potential solution).
- The 3GPP project plan was updated to reflect the CN IETF dependencies.
- Monthly discussions were promulgated, with the ultimate goal of utilizing the Internet functionality, when possible, in the 3G wireless network.
- SIP's was a slow standardization process but the group was split into SIP and Session Initiation Protocol Project INvestiGation (SIPPING) groups, to increase its efficiency.
- An IETF IPNg/3GPP ad-hoc was organized and the SA WG2 coordinated and presented the 3GPP system architecture in order to improve the knowledge of the 3GPP Project.
- The IETF/3GPP collaboration process had passed to the final stage, and the review of the IETF-3GPP collaboration draft was finalized in April.

AAA/Diameter

Initially the SA5 approved a liaison statement in February (TdocS5-010127) summarizing the release 4 AAA requirements. Presently, all fourteen salient differences between the Charging rapporteur group protocol requirements and the main IETF AAA protocol requirement document are incorporated into the RFC 2989.

The Diameter Based Protocol is intended to provide an AAA framework for Mobile-IP, NASREQ, and ROAMOPS (Roaming Operations). This document, <**draft-ietf-aaa-diameter-05.txt**>, issued in June 01, specifies the "message format, transport, error reporting and security services to be used by all Diameter applications and must be supported by all Diameter applications and must be supported by all Diameter implementations."

The following timeline is currently proposed for this document:
- Service requirements finalized by 1st of July
- IESG review scheduled in September

- Interoperability test event - 1st week in October.

Considering the current stage of development, and with efficient coordination, it appears possible to have the first AAA charging protocol version completed within the Release 5 timeframe.
Note: <**draft-ietf-aaa-diameter-02.txt**> was issued in April.

SIP

IETF SIP Group Reorganization

This was a slow standardization process but the group was split into SIP and Session Initiation Protocol Project INvestiGation (SIPPING) groups, to increase its efficiency.

The Session Initiation Protocol (SIP) working group is chartered to continue the development of SIP, currently specified as proposed standard RFC 2543. The SIP working group will concentrate on the specification of SIP and its extensions (i.e. general-purpose requirements for changes to SIP provided by other working groups, including the SIPPING working group).

The SIPPING working group is chartered to document the application of SIP to certain domain tasks and to develop requirements for the changes or extensions to SIP needed to accomplish those tasks. The SIPPING working group will concentrate on the frameworks, requirements, and practices related to SIP and its extensions, and will not specify changes or extensions to SIP
The specific deliverables of the group are:

- Analyze the requirements for application of SIP to several different tasks (e.g. SIMPLE that is using SIP for instant messaging and presence).
- Support seamless inter-working between ISUP and SIP for the phone-to-phone call without making SIP overly complex for the calls between the PSTN/ISDN and IP phone.

Tracked Documents

The specific deliverables of the SIP and SIPPING groups that are related with the existing 3GPP working items:

1. The RFC 2543 is the only work on SIP that can be referenced. The main SIP document, RFC 2543, is currently being updated by a new id **draft-ietf-sip-rfc2543bis-03.txt**, revised at the end of May.
   The following timeline is currently proposed for this document:

   - Draft-03 ready in late May
   - Draft-04 should be ready in September
   - Last call process is scheduled for 6th of October
   - Conclusion collections and integration by 3rd of November
   - Final document ready - 24 of November
   - Available for IESG review at the 15 of December

Note: for some subjects such as codecs negotiation the solutions are very broad. Additional work may be needed to align the draft with the current CRs.

2. Reliability of provisional responses in SIP (**draft-ietf-sip-100rel-02.txt**). This document defines a new PRACK method (i.e. new header fields) and it does not have dependencies on 2543bis. The following timeline is currently proposed for this document:

   - Ready for the IESG review by 15 of September.

- Final document ready - 15th of October
- Document submission - 1s half of November

3. Integration of resource management and SIP. This document **(draft-sip-manyfolks-resource-00.txt)** discusses how establishment of QoS and security procedures can be made a precondition to sessions initiated by the SIP. It proposes an extension to SIP to add a new COMET method. The current version does not have dependencies on 2543bis. The following timeline is proposed for this document:

    - Ready for IESG review by 15th of September
    - Final document ready - 15th of October
    - Document submission - 1st half of November

4. SIP extensions for caller identity and privacy (**draft-ietf-sip-privacy-00.txt**). This document defines new Anonymity and Remote-Party-Id headers and the extensions that allow the parties to be identified either by name or by type, the latter of which can be used to identify some group of callers and callees. There are no known issues, and only few comments were received. The following timeline is currently proposed for this document:

    - Ready for the IESG review by 1st of September.

    Note: This document was not on the SIP/SIPPING calendar, but this matter was remedied. However, 3GPP should be proactive to prioritize the service requirements (e.g. call trace, called ID, emergency calls).

5. SIP extensions for media authorization **(draft-ietf-sip-call-auth-00.txt**). This document defines a new Media-Authorization header. It was accepted as a SIP WG item in December and it depends of 2543bis draft as uses 183 response.

    - This draft will be included in the SIP group calendar
    - It was ready for the last call in April
    - Its schedule will be reviewed at the next meeting.

6. SIP event notification extension is defined by the **draft-ietf-sip-subscriber-notify-03.txt**. The purpose of this extension is to provide a generic framework for the SIP nodes to request notification from remote nodes. It can be used for many purposes such as network management. This document expires in August and the document should be:

    - Ready for the IESG review by 7th of October

7. SIP calls control transfer. The **draft-ietf-sip-cc-transfer-04.txt** defines the SIP extensions within the call control. The following timeline is proposed for this document:

    - Ready for the IESG review by 1st of August

8. The encapsulation of ISUP over SIP messages using MIME format has become a standard in the IETF. This document defines the MIME types for application/ISUP and application/QSIG objects for use in SIP applications.

9. The mapping of ISUP-over-SIP. The **draft-ietf-sip-isup-00.txt** provides the intelligence of routing as well as service creation capability over the SIP network in the following call scenarios: PSTN phone-to-PSTN phone via SIP network, PSTN phone-to SIP phone, SIP phone-to PSTN phone. The timeline for this document is:

- Ready for the IESG review in 1st of September.

10. SIP-H.232 Interworking. The timeline for the **draft-agrawal-sip-h323-interworking-reqs-02.txt** is as follows:

- Ready for the IESG review in July

11. ISUP to SIP mapping. The **draft-ietf-sip-isup-header-00.txt** describes a way to perform the mapping between the two signaling protocols.

    - Not scheduled

New possible dependencies

The following documents are temporarily referenced in 3GPP documents (e.g. TS 24.229 - IP Multimedia Call Control Protocol based on SIP and SDP):

1. Draft-ietf-sip-callerprefs-03.txt; it describes the caller preferences and caller capabilities

2. Draft-ietf-sip-serverfeatures-04.txt; it describes the service features supported by the header

3. Draft-ietf-sip-state-01.txt; it describes the SIP extensions for supporting distributed call state

4. Draft-ietf-sip-session-timer-04.txt;

5. RFC 2327 (April 1998): "SDP: Session Description Protocol".

Ipv6

An IETF/3GPP meeting was held at the end of May in Redmond. The 3GPP members presented an introduction to the 3GPP architecture, the concepts of the UMTS packet domain, the Ipv6 requirements on the CN and UE systems and the possible Ipv4/Ipv6 transition methods. As

examples, three alternatives were discussed: dual stack, NAT/PT and tunneling (see 23.221).

It was proposed to form an IETF/3GPP design team with specific interests on some questions raised at the meeting (e.g. the PDP context support of multiple IP addresses). The general idea is that the design team will get together, review the issues and submit their work. They may correspond on the main list, on private lists, or however they need to get their work done. This is not a formal structure.

Possible New Dependencies – Observations

1. XML encoding for SMS messages. The <**draft-koonen-sms-xml-01.txt**>, posted on April 18, 2001, presents an encoding and simple protocol for describing and submitting SMS messages over the Internet (i.e. between service providers and SMS gateways)
2. SIP security. SIP has the built-in mechanism for the end-to-end security using PGP and related things. However, the crossing of the firewalls and NATs by application layer protocols like SIP is still a problem. The security has become the main focus especially for SIP. In this context, at the last IETF meeting a tutorial on security (http://jis.mit.edu/sectutorial) was presented.

3. At the last IETF meeting an unofficial BOF was conducted among the SIP WG participants to formulate the features needed for SIP emergency services. An URL (http://www.cs.columbia.edu/sip/emergency.html) has been established for this particular purpose.
4. Two documents relating to the smart cards were listed in January but there is no progress to report:
   - The "draft-guthery-tcp7816-01.txt draft describes the transport of TCP and UDP packets over the IP layer of ISO 7816 integrated circuit ("smart") cards with particular attention to header compression. It expires in July and there is no working item defined in Release 5.
   - The "draft-guthery-ip7816-01.txt" draft describes the transport of IP datagrams and ARP messages over the ISO 7816 link layer of integrated circuit ("smart") cards. It expires in July and there is no working item related with this item in Release 5.
5. AVT WG supports the real time protocol (RTP) to enhance its capabilities for codecs with improved performance that can be used by SIP.
6. SIMPLE WG supports to enhance SIP capabilities for integration with the real-time instant messaging (IM) and presence.

Recommendations

1. 3GPP individual members are encouraged to be active within the IETF via mailing lists and participate in the various studies and answering the various questions posed by the IETF ADs. For important drafts (e.g. 2543bis) we should start to assign people to read-proof sections and read the entire document(s) and verify that their current organization needs no major changes.
2. We will continue to solicit help from IETF to accelerate the standardization process in the areas were strong 3GPP release 5 project plan exists (i.e. transport - SIP)
3. Proactively, I will continue the periodic dialogue with ADs and build the awareness of the areas of importance for Release 4/5 fulfillment (e.g. SIP, QoS, Ipv6).
4. Understand the scope of some new IETF efforts and their relation with relevant areas to 3GPP (e.g. presence services and SIP, IPv6, local area)
   - Presence and Instant Messaging Protocol (prim) group defines a protocol compliant with CPIM (Common Profile for Instant Messaging)
   - SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE) group investigates the ongoing work towards the standardization of SIP for presence as a transfer protocol supported within the CPIM framework
   - Site Multihoming in Ipv6 - the group aims is to discuss the multihoming approach.

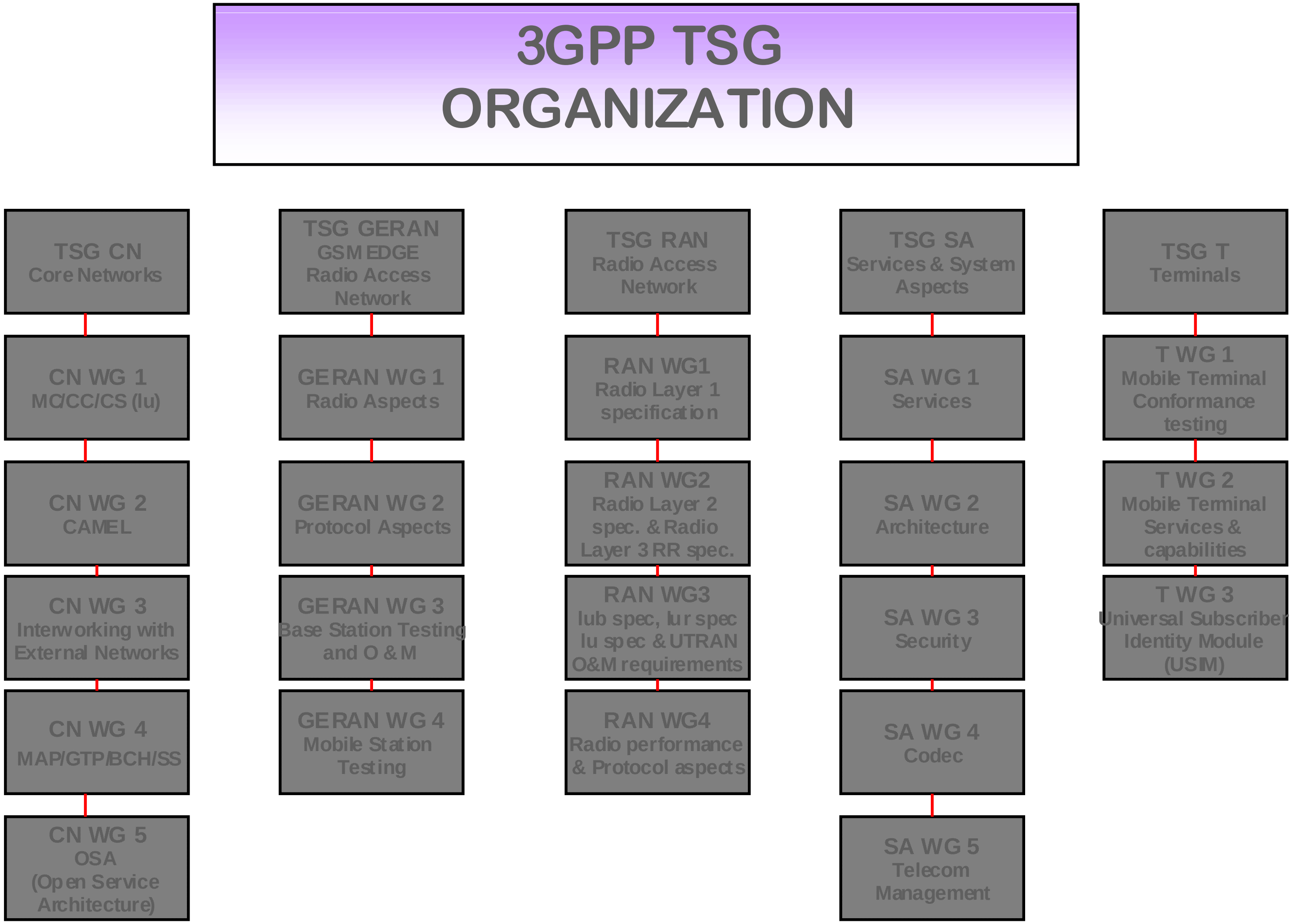
UMTS and IPv6

Presentation Outline

- Overview of 3GPP
- Introduction to 3GPP architecture
- Concepts of the UMTS packet domain
- IPv6 in UMTS
- Summary

Overview of 3GPP

Overview of 3GPP (1/2)

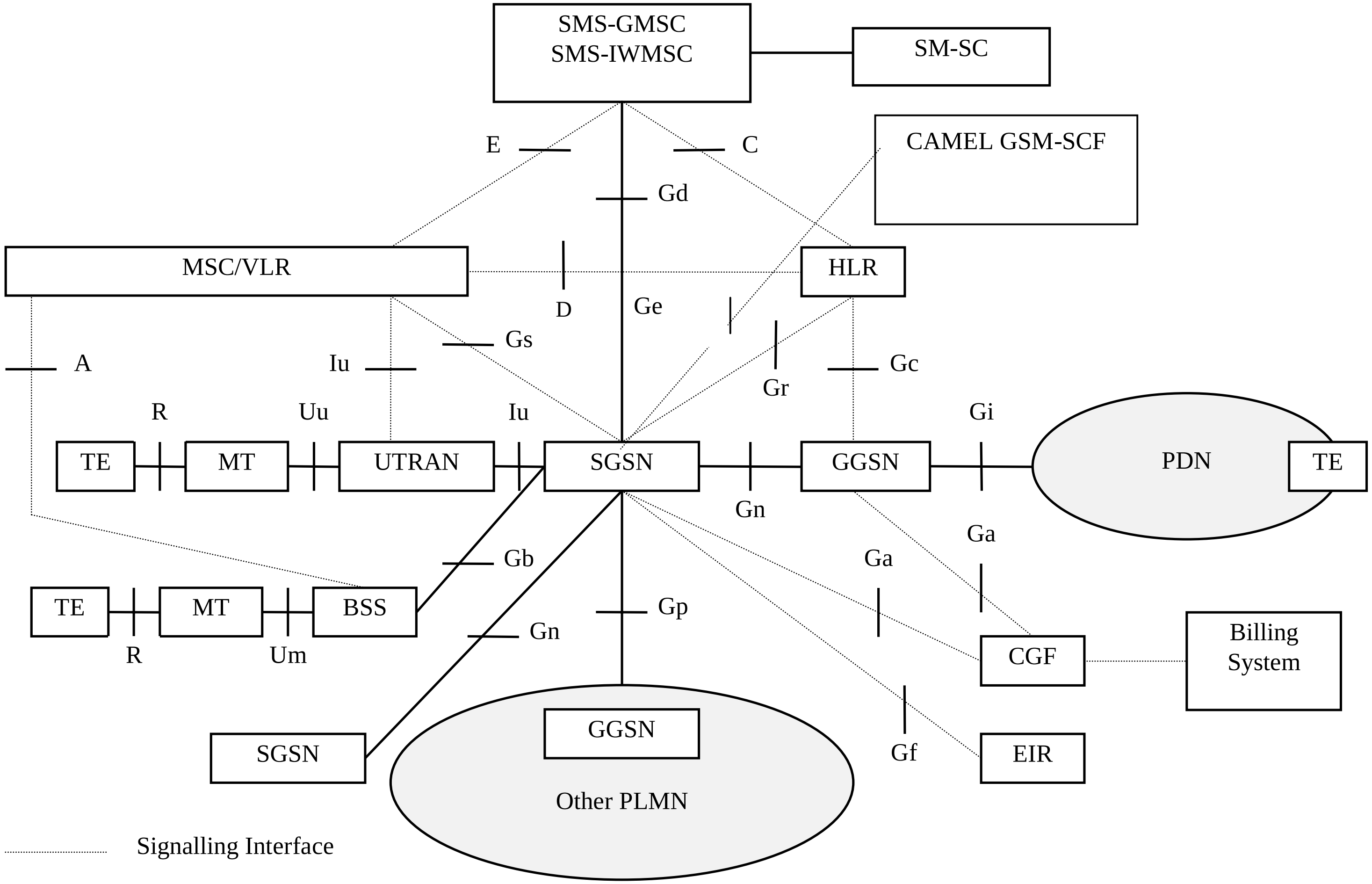


Overview of 3GPP (2/2)

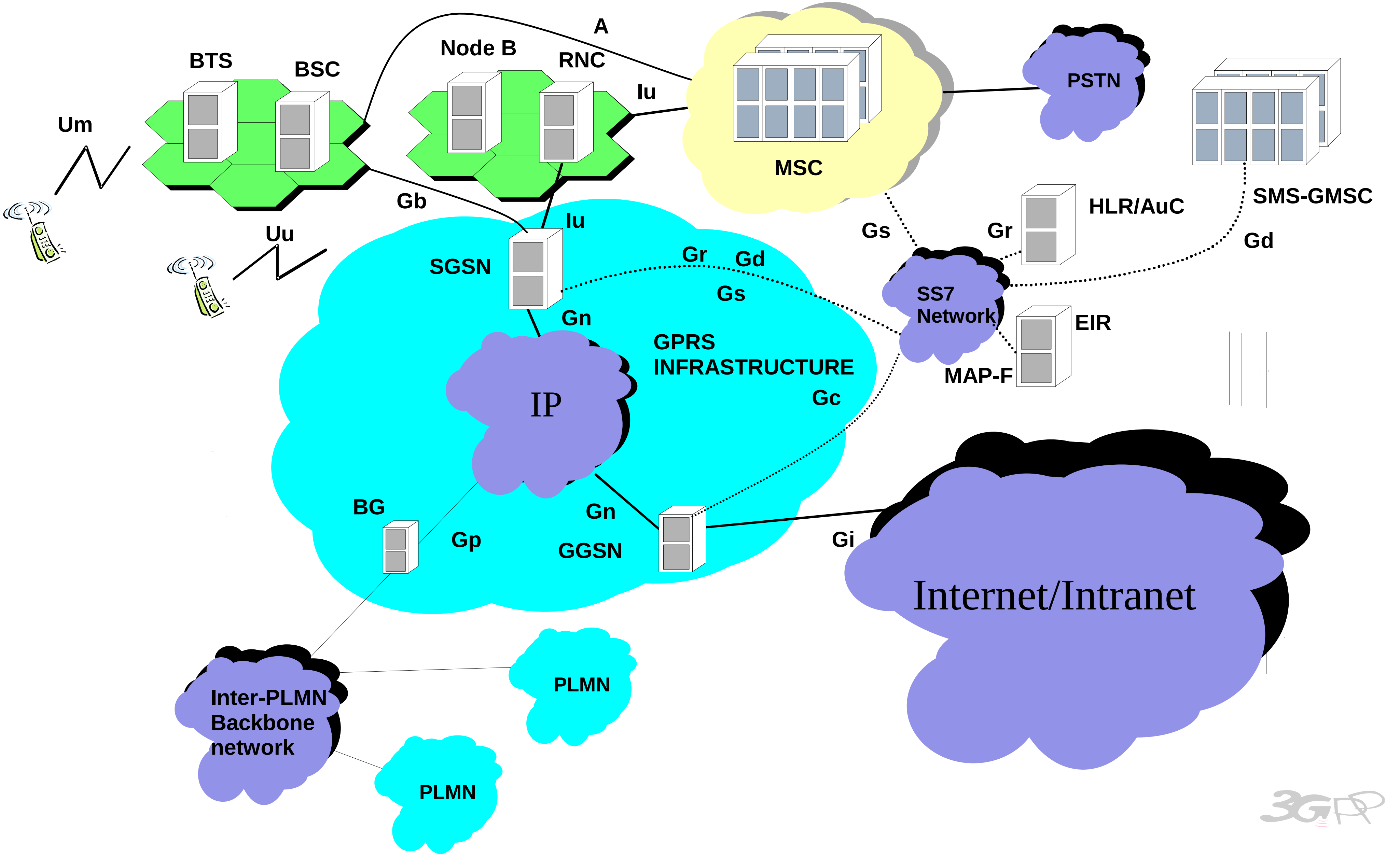
- Technical Work Done in WGs
- Meetings
  - As Necessary
  - Decision through Consensus or Voting
  - Most of the Work Done in Meetings
- Deliverables
  - Technical Reports/Technical Specifications
  - Approval by Consensus or Vote
  - Change Control When Sufficiently Stable
- Inter-WG Coordination
  - In TSGs
  - Information Exchange through Liaison Statements
- Standards
  - Releases

The UMTS Architecture

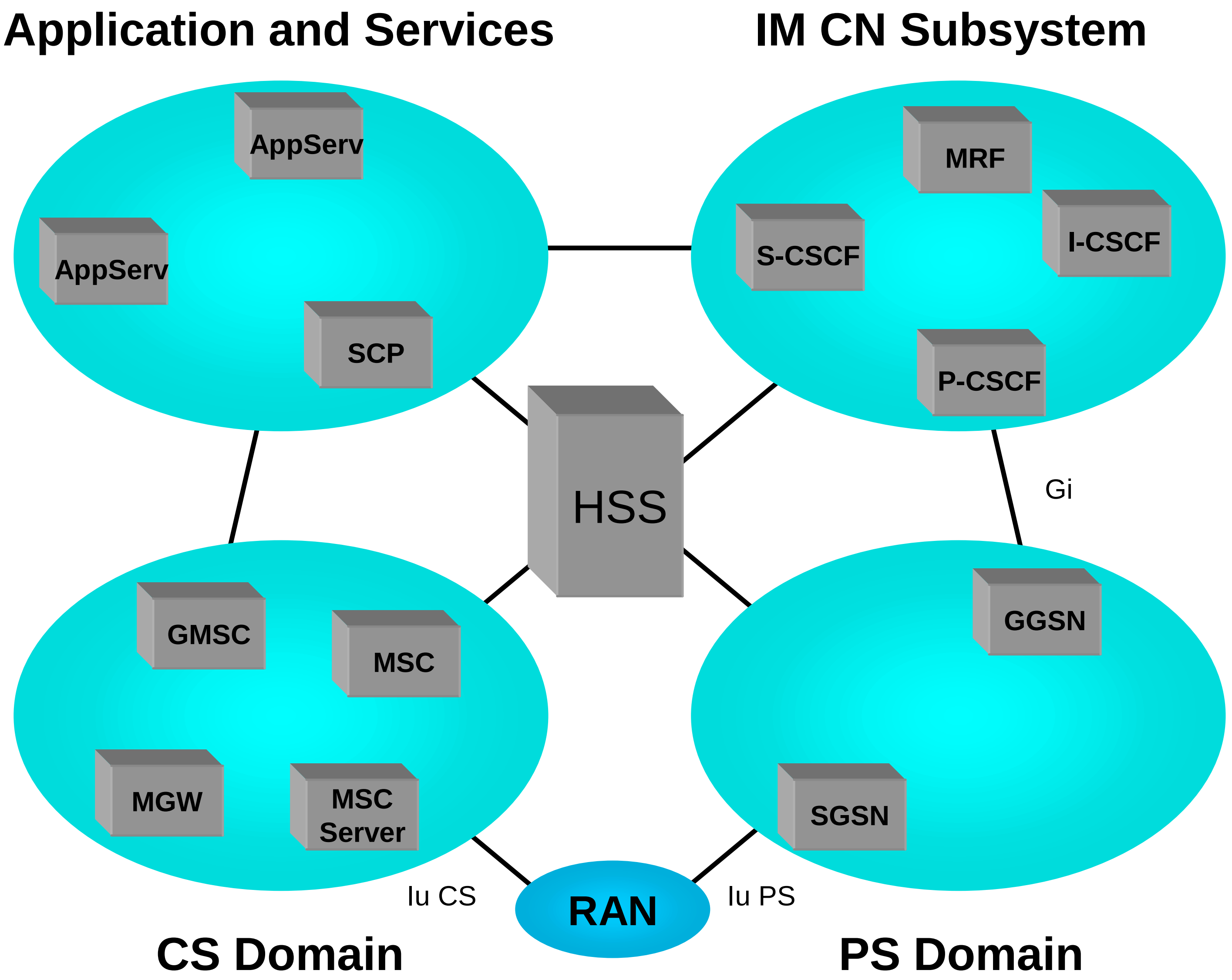
R’99 UMTS/GPRS Architecture



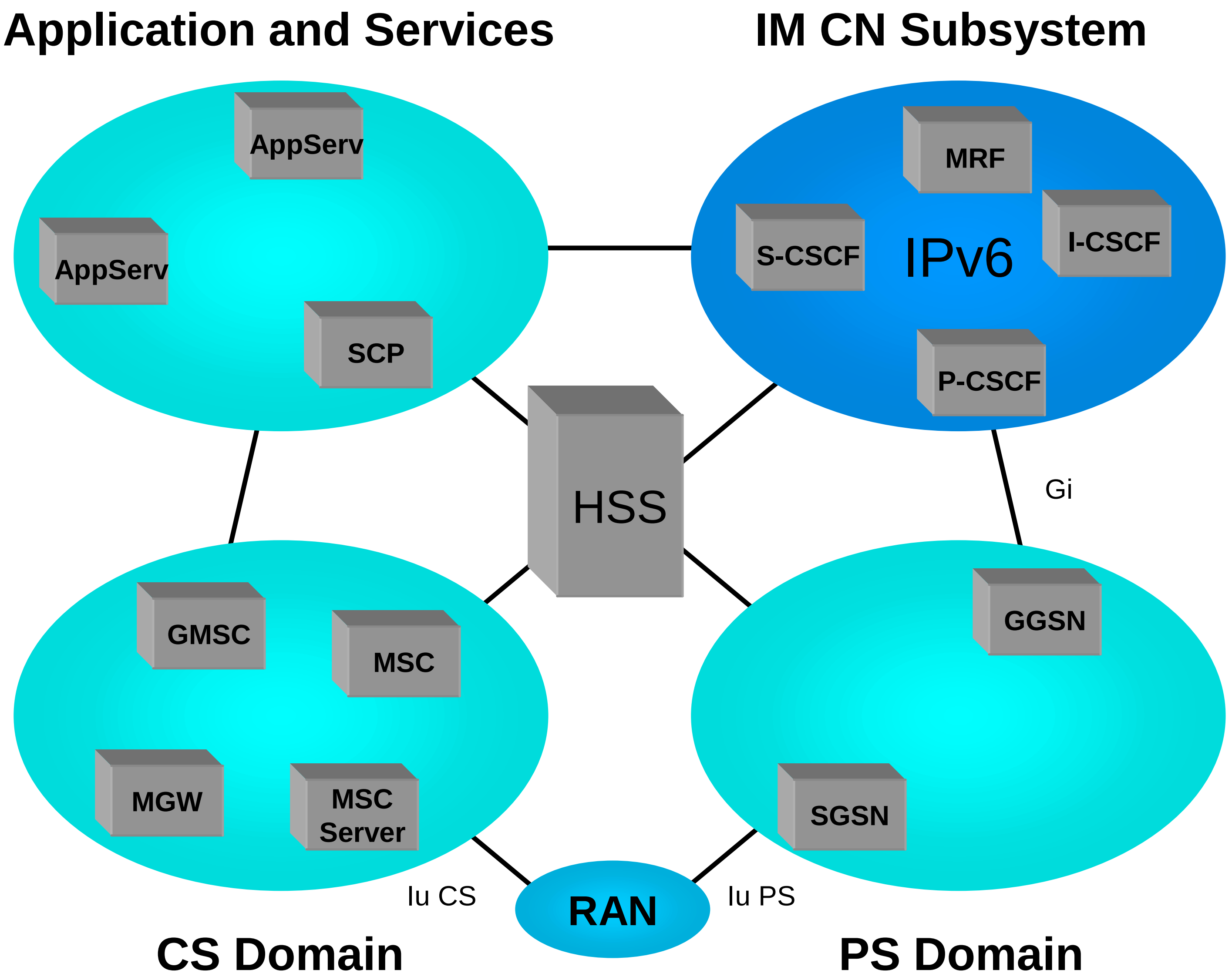
R’99 UMTS/GPRS Architecture



Release 4/5 Architecture

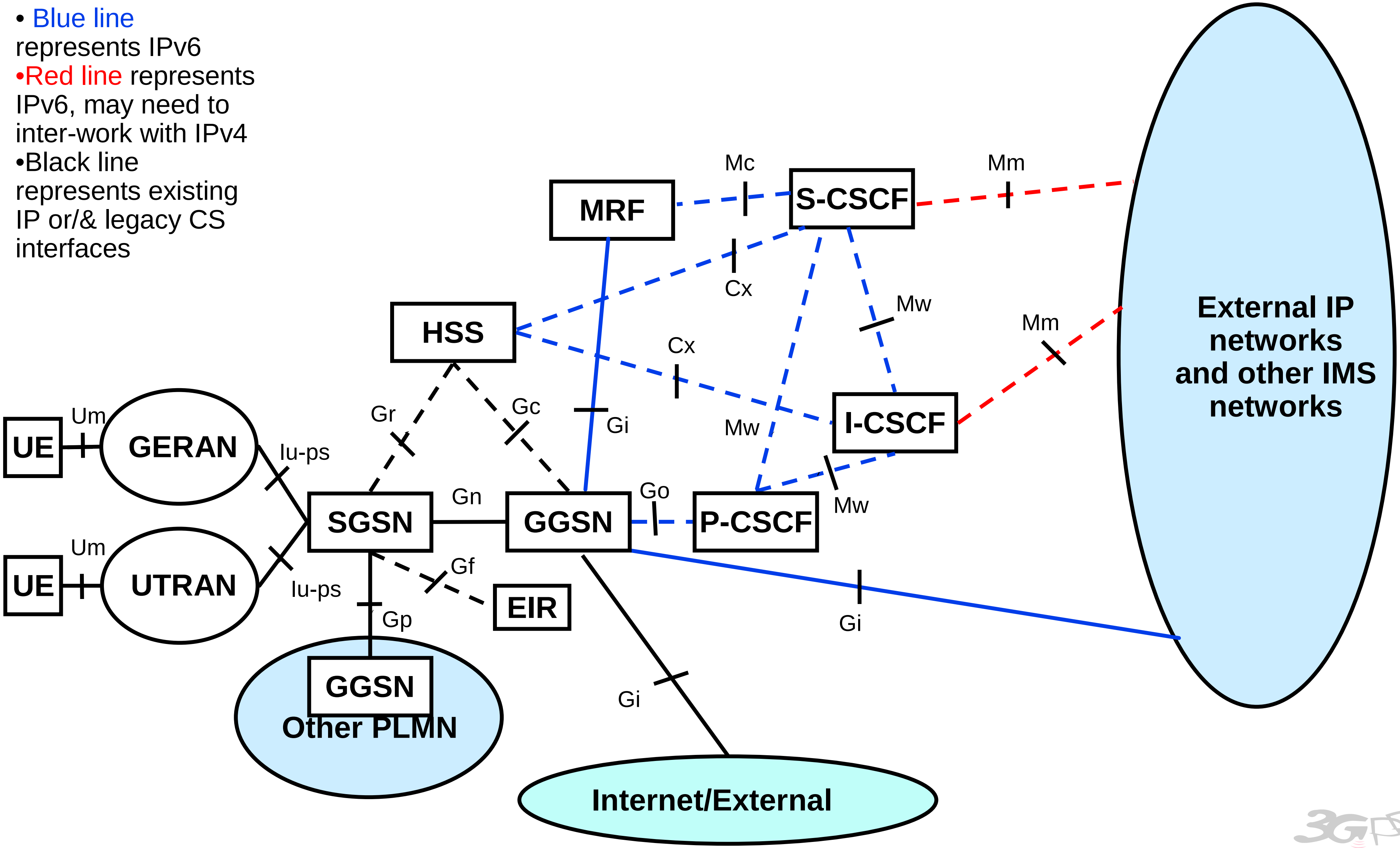


Release 4/5 Architecture

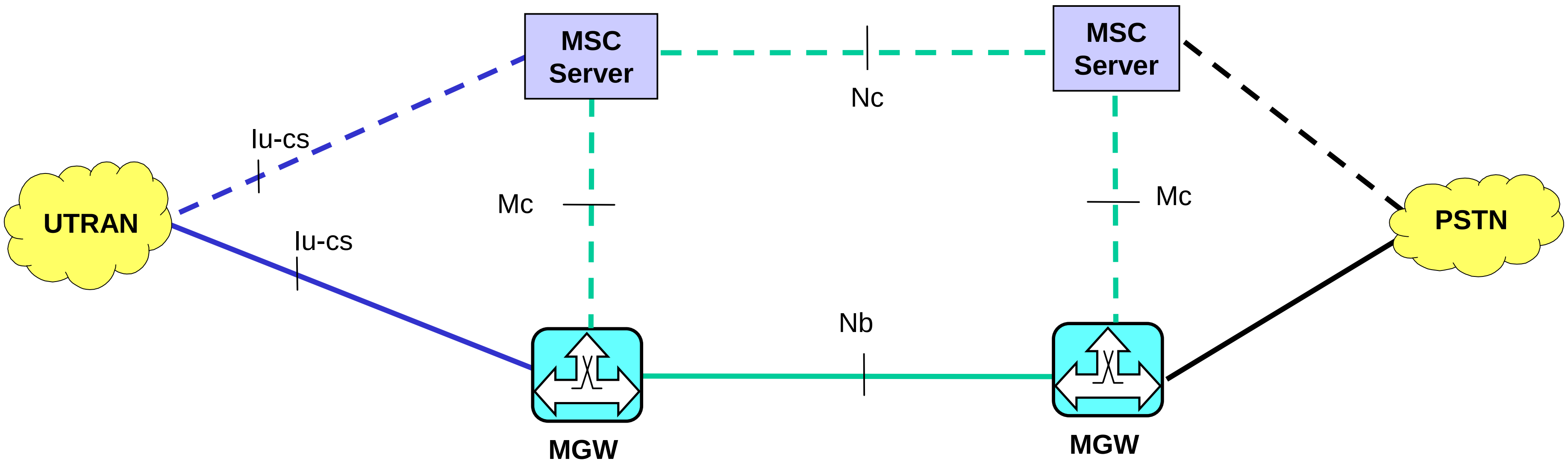


Simplified model for IP Multimedia

- Blue line represents IPv6
- Red line represents IPv6, may need to inter-work with IPv4
- Black line represents existing IP or/& legacy CS interfaces

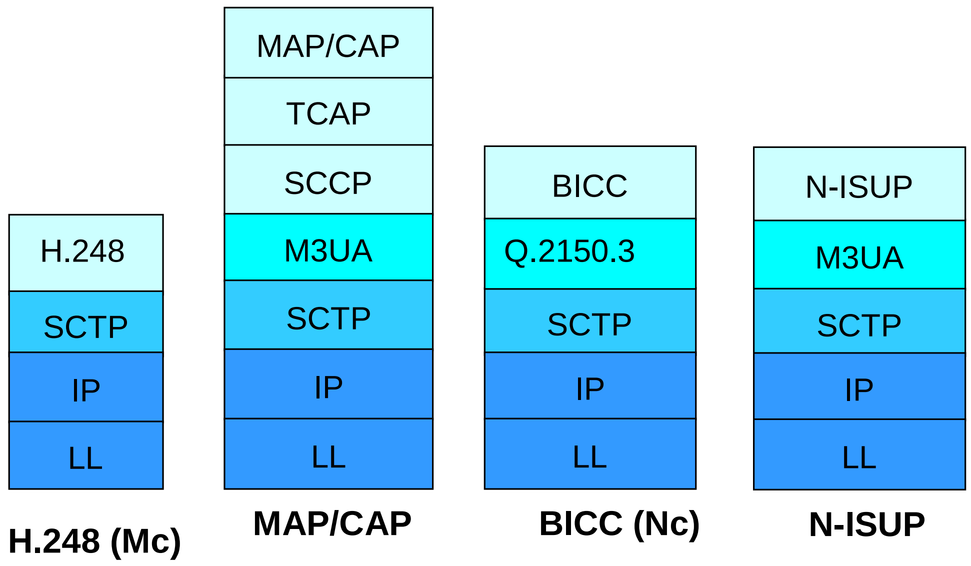


CS Domain: Signaling & User Plane

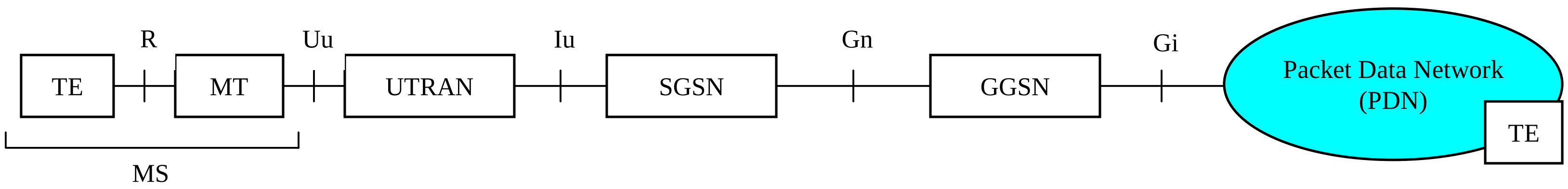


This bearer independent architecture makes possible to use IP transport

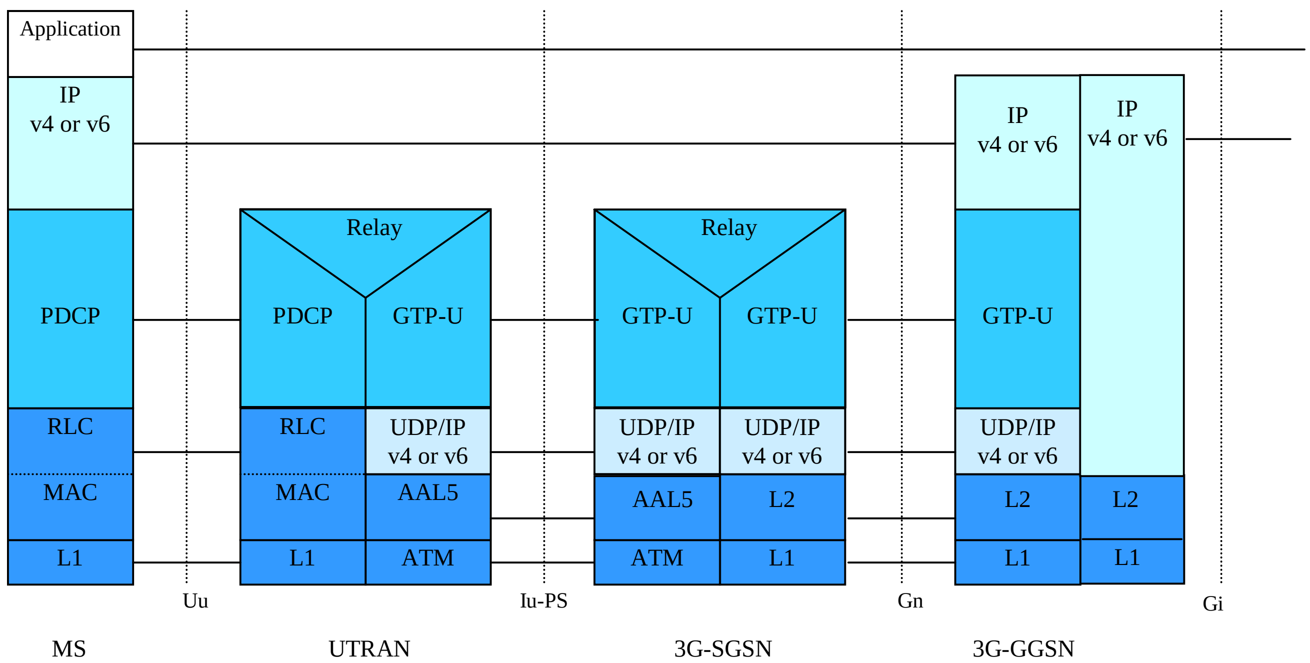
CS domain protocol stack using IP transport option



Simplified PS Domain Architecture

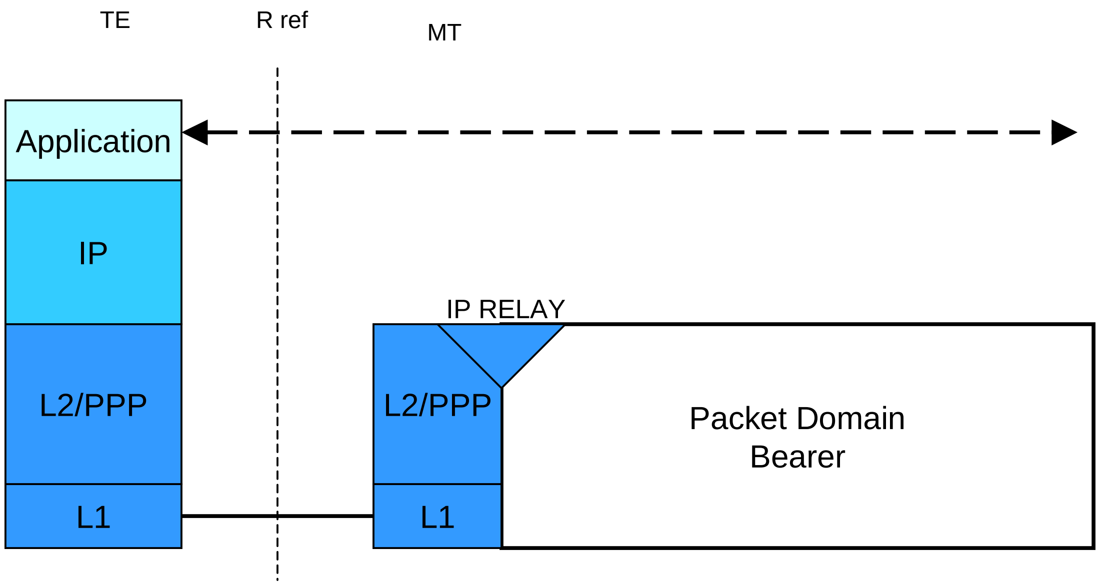


PS Domain User Plane protocol stack



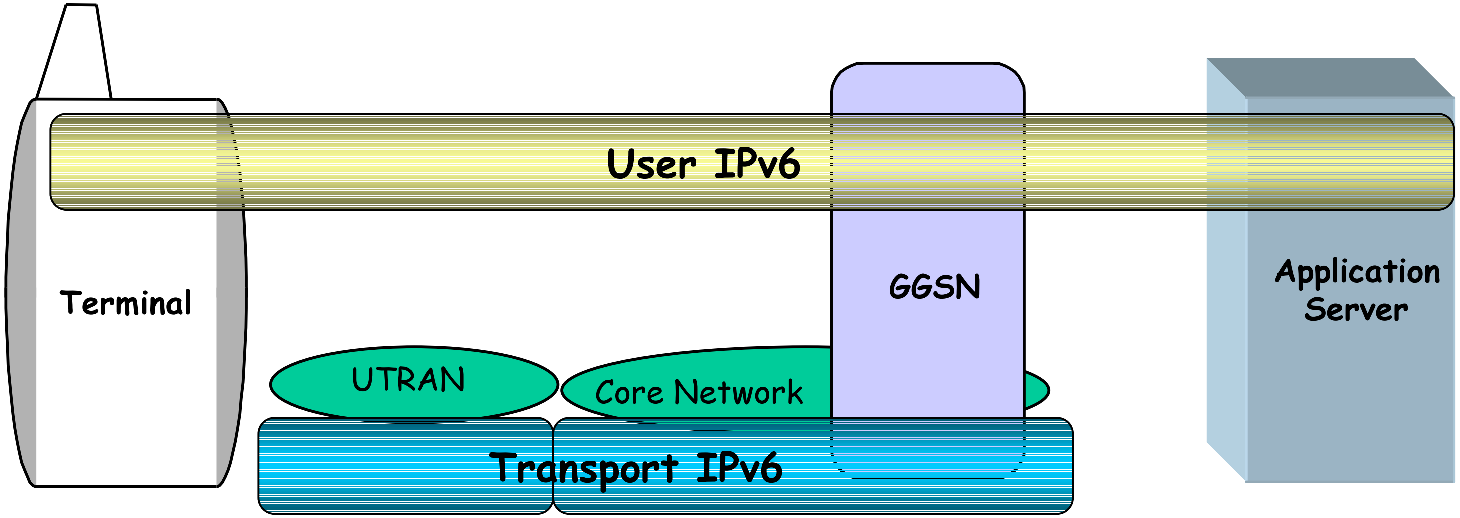
MT-TE Configuration

IP based services



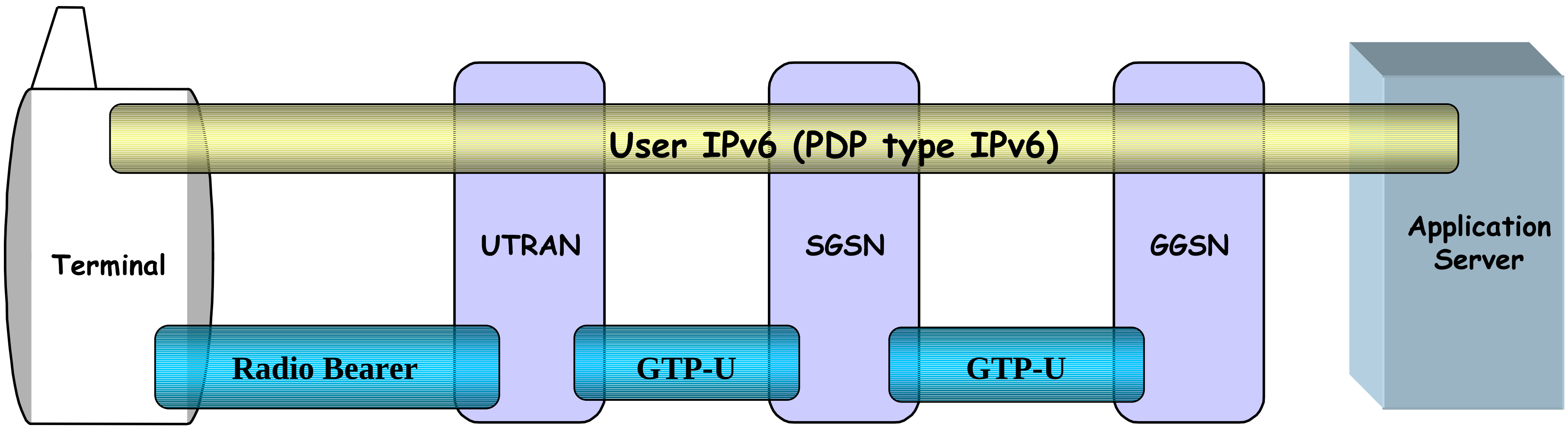
Note: MT and TE can be physically separated or physically co-located

User plane vs transport plane



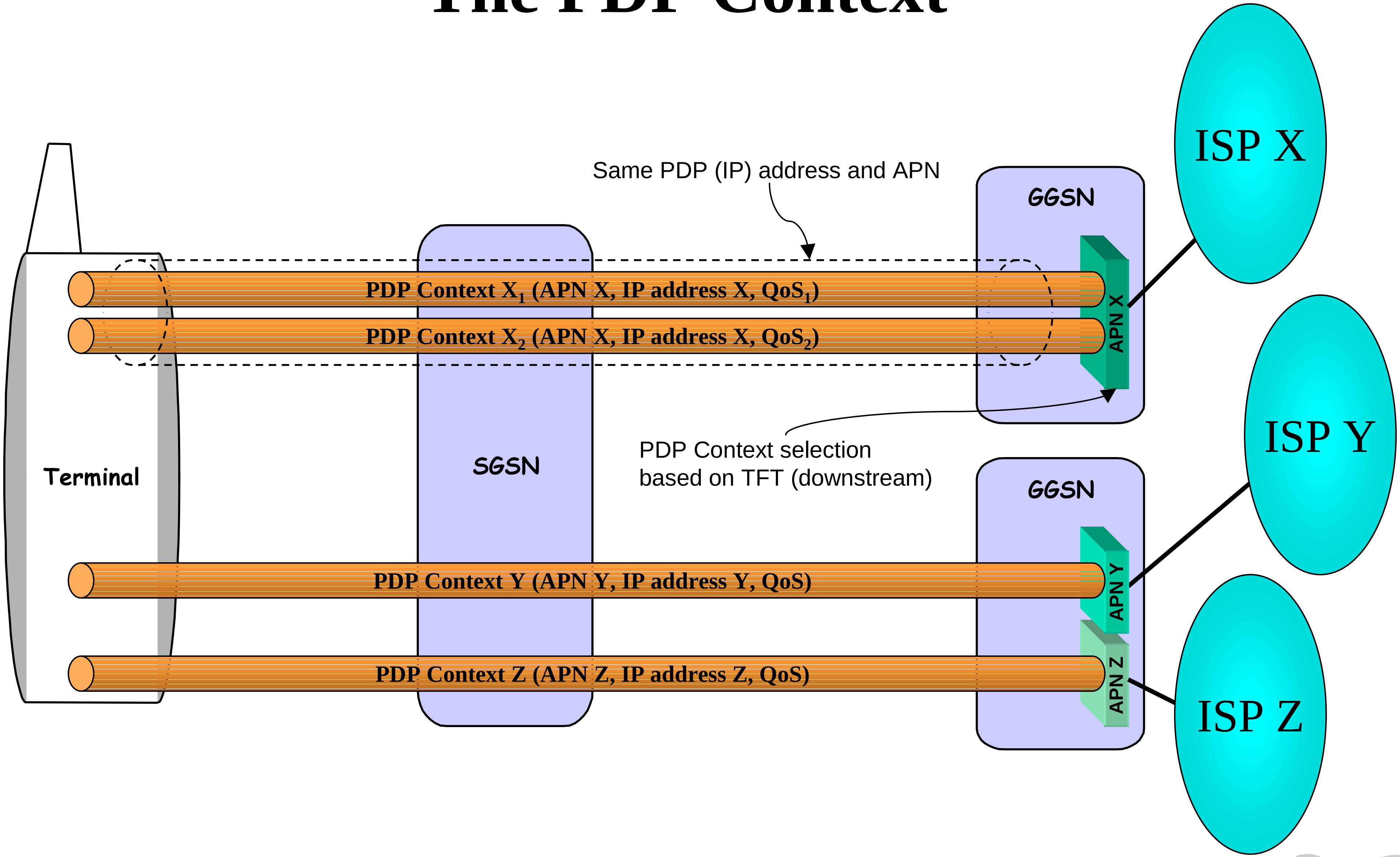
- User and transport planes are completely independent, i.e. the transport plane can run on a different IP version than the user plane
- UTRAN and Core Network transport can also run on different IP versions

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 the IP level.

The PDP Context



The PDP CONTEXT

When an MS attaches to the Network, the SGSN creates a Mobility Management context containing information pertaining to e.g., mobility and security for the MS.

At PDP Context Activation (PDP - Packet Data Protocol), the SGSN and GGSN create a PDP context, containing information about the session (e.g. IP address, QoS, routing information , etc.),

Note: Each Subscriber may activate several PDP Contexts towards the same or different GGSNs. When activated towards the same GGSN, they can use the same or different IP addresses.

The Access Point Name - APN

The APN is a logical name referring to a GGSN. The APN also identifies an external network.

The syntax of the APN corresponds to a fully qualified name.

At PDP context activation, the SGSN performs a DNS query to find out the GGSN(s) serving the APN requested by the terminal.

The DNS response contains a list of GGSN addresses from which the SGSN selects one address in a round-robin fashion (for this APN).

Traffic Flow Template (TFT)

A TFT is a packet filter allowing the GGSN to classify packets received from the external network into the proper PDP context.

A TFT consists of a set of packet filters, each containing a combination of the following attributes:

- Source Address and Subnet Mask
- Destination Port Range
- Source Port Range
- IPsec Security Parameter Index (SPI)
- Type of Service (TOS) (IPv4) / Traffic Class (IPv6) and Mask
- Flow Label (IPv6)

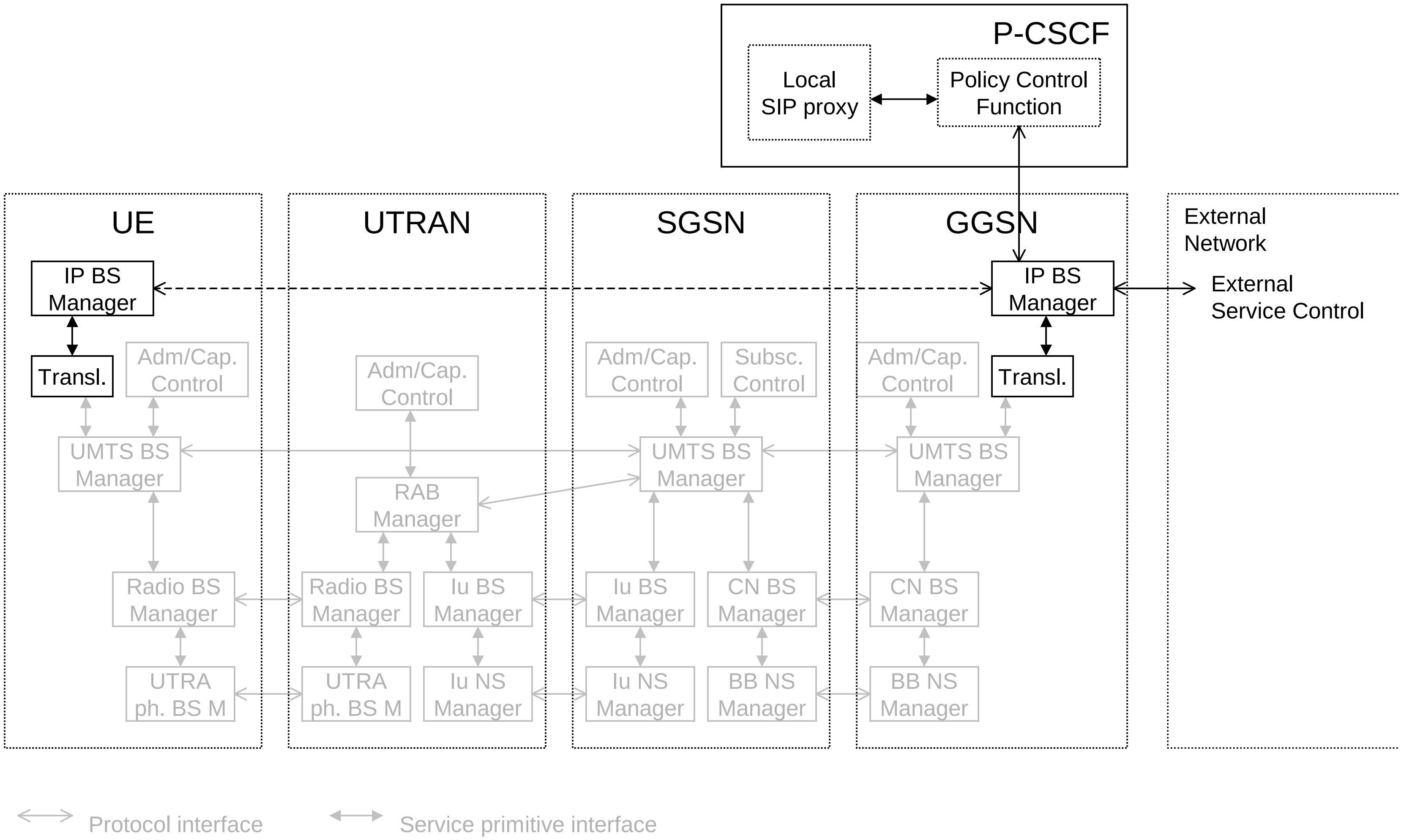
GPRS Tunneling Protocol

GTP is a simple tunneling protocol based on UDP/IP, used both in GSM/GPRS and UMTS.

A GTP tunnel is identified at each end by a Tunnel Endpoint Identifier (TEID)

For every MS, one GTP-C tunnel is established for signalling and a number of GTP-U tunnels, one per PDP context (i.e. session), are established for user traffic.

QoS Management Functions in UMTS



IP BS Manager

- is used to control the external IP bearer service to provide IP QoS end-to-end.
- communicates with the UMTS BS manager through the translation function.
- uses standard IP mechanisms to manage the IP bearer service.
- may exist both in the UE and the Gateway node, and it is possible that these IP BS Managers communicate directly with each other by using relevant signalling protocols, e.g., RSVP
- is the policy enforcement point for Service-based Local Policy control

Policy Control Function (PCF)

- is a logical entity that is co-located with the P-CSCF (the interface between the P-CSCF and PCF is not standardized in Release 5)
- is a logical policy decision element which uses standard IP mechanisms to implement Service-based Local Policy in the bearer level
- enables coordination between events in the SIP session level and resource management in the bearer level
- makes policy decisions based on information obtained from the P-CSCF
- has a protocol interface with GGSN (Go interface) which supports the transfer of information and policy decisions between the policy decision point and the IP BS Manager in the GGSN (following COPS framework)

IP BS Manager capability in the UE and GGSN

Table 1: IP BS Manager capability in the UE and GGSN

Capability	UE	GGSN
DiffServ Edge Function	Optional	Required
RSVP/Intserv	Optional	Optional
IP Policy Enforcement Point	Optional	Required (*)

(*) Although the capability of IP policy enforcement is required within the GGSN, the control of IP policy through the GGSN is a network operator choice.

IPv6 Details

IPv6 History in UMTS

- IPv6 in the 3GPP standards
  - User plane: PDP Type IPv6 introduced in GPRS R'97
  - Transport plane: IPv6 is optional
  - UTRAN: IP transport study is being conducted right now
  - IMS: The IP Multimedia Core Network Subsystem has been standardized to be based on the following IPv6 support:

    - The architecture shall make optimum use of IPv6.

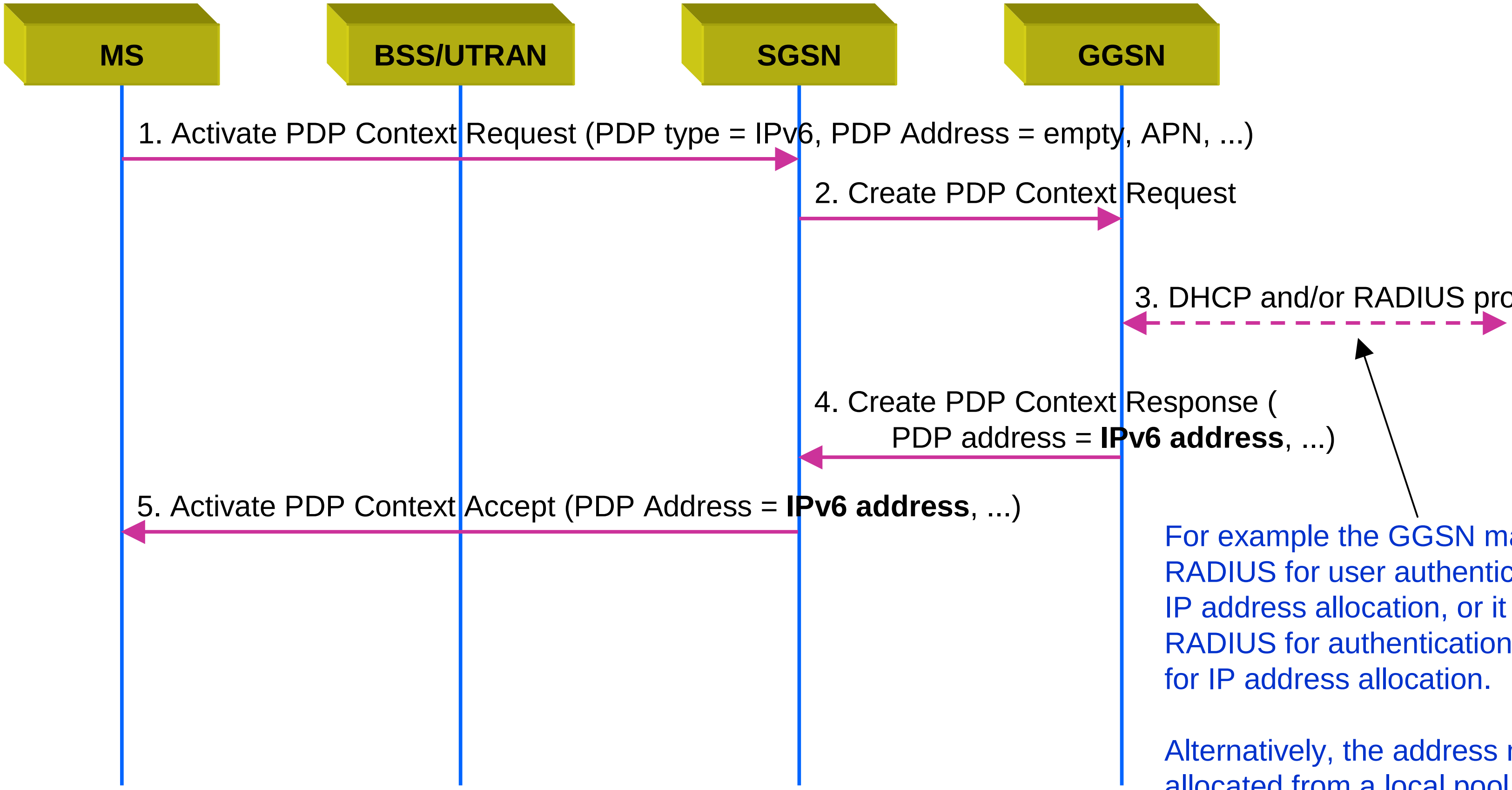
    - The IM CN subsystem shall exclusively support IPv6.

    - The UE shall exclusively support IPv6 for the connection to services provided by the IM CN subsystem.

IPv6 Address Allocation Methods

- Stateless Address Autoconfiguration
  - Introduced in GPRS R'99
- Stateful Address Autoconfiguration
  - DHCPv6 client in the terminal
  - Requires DHCPv6 relay agent in the GGSN
- GPRS-specific Address Configuration
  - Static Address Configuration
    - The MS provides its statically configured IPv6 address at PDP context activation
  - Dynamic Address Allocation
    - The IPv6 address is provided by the GGSN at PDP context activation

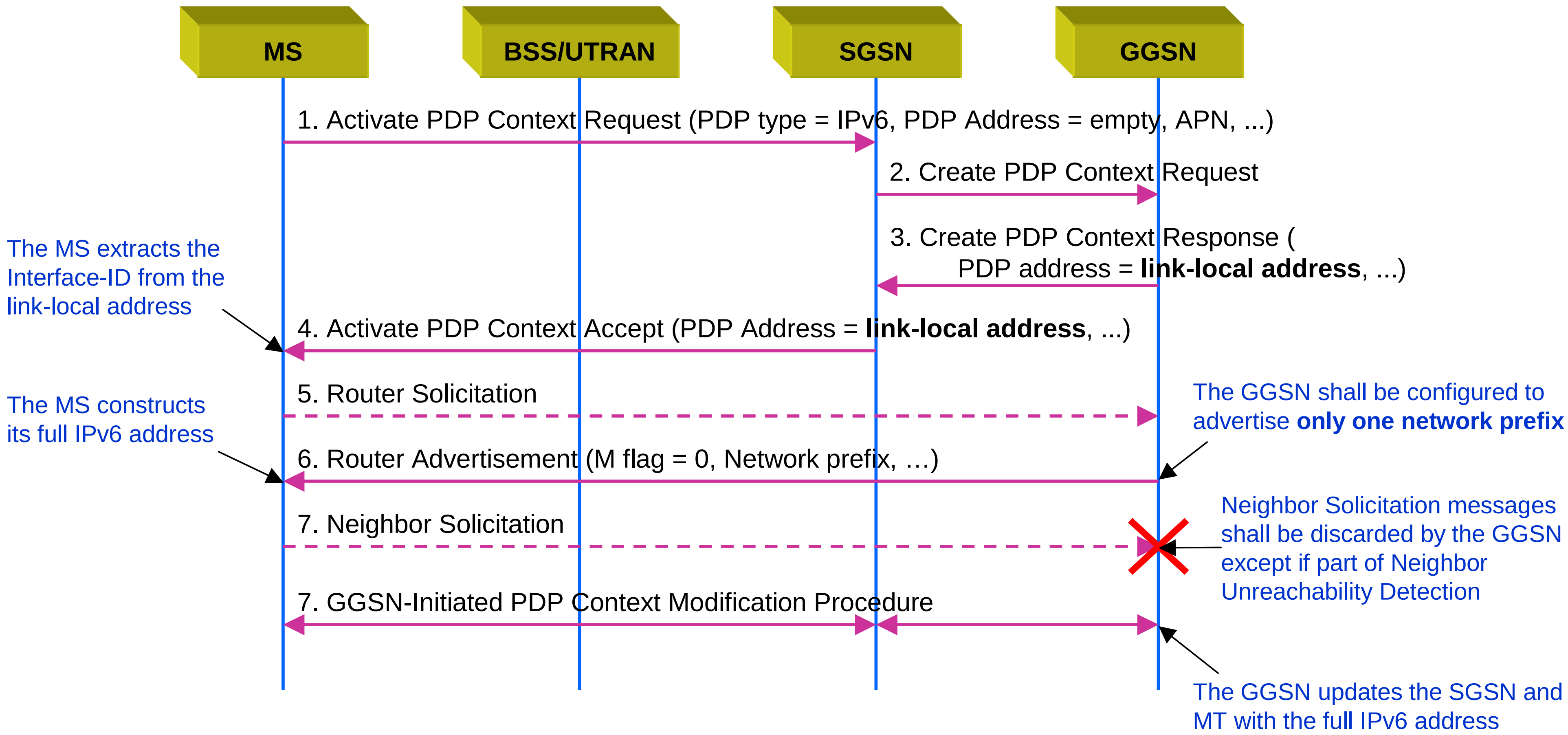
Dynamic Address Allocation in UMTS/GPRS



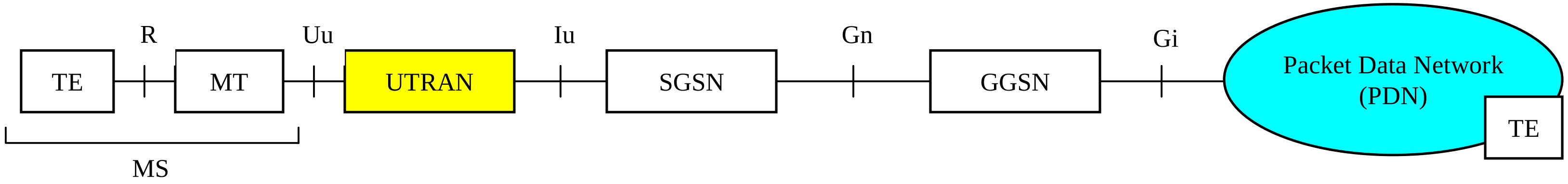
For example the GGSN may use RADIUS for user authentication and IP address allocation, or it may use RADIUS for authentication and DHCP for IP address allocation.

Alternatively, the address may be allocated from a local pool of addresses in the GGSN.

Stateless Address Autoconfiguration in UMTS/GPRS



Header Compression

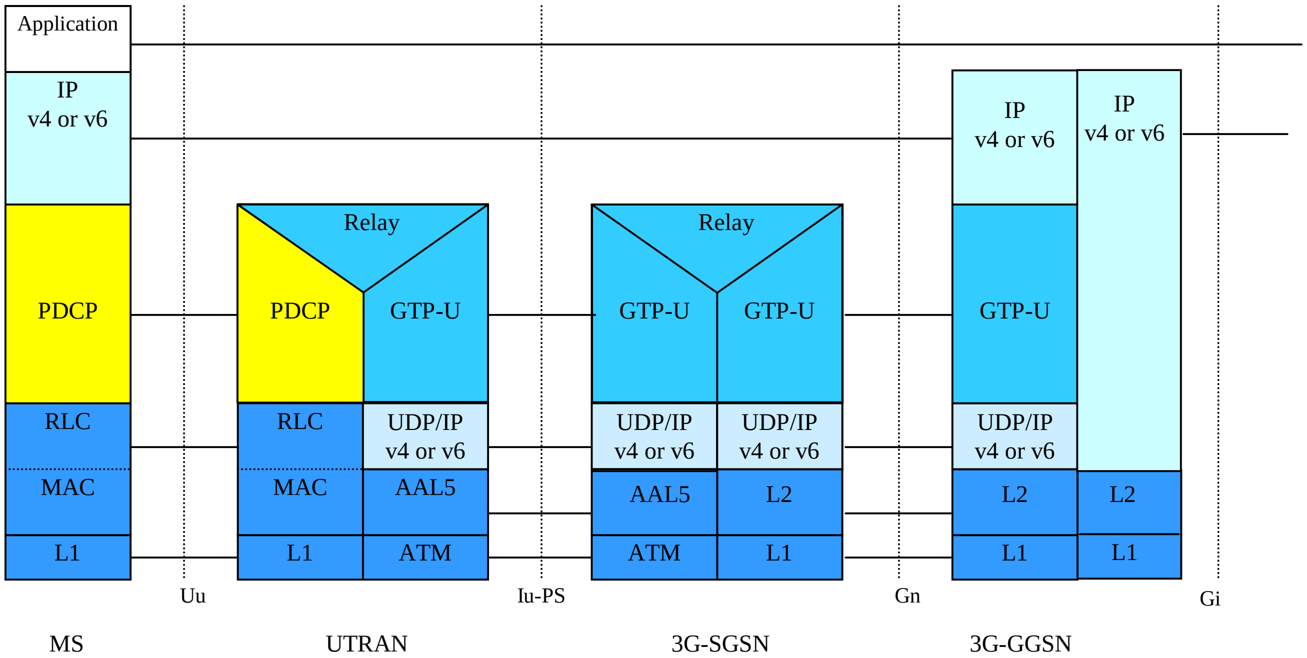


PS Domain User Plane protocol stack

Header Compression:

•RFC2507

•RFC…



IPv4/IPv6 Transition

Text in 23.221 shows examples of transition:

- Dual Stack
- NAT/PT
- Tunneling

These are only examples to show how transition could be done.

They are not mandatory to implement/deploy.

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