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Technical Report

3rd Generation Partnership Project;
Technical Specification Group Services and Systems Aspects;
Evolution of the GSM platform towards UMTS
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Contents

Foreword

This Technical Report has been produced by the 3GPP.

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of this TS, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version ~~3~~x.y.z

where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

3 Indicates TSG approved document under change control.

y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

z the third digit is incremented when editorial only changes have been incorporated in the specification;

1 Scope

The present document covers issues related to the evolution of the GSM platform towards UMTS with the overall goal of fulfilling the UMTS service requirements, the support of the UMTS role model, support of roaming and support of new functionality, signalling systems and interfaces.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.
- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.

[1] ~~ETSI TC-SMG UMTS~~ TS 22~~-~~.001: "Services Principles"

[2] ~~ETSI TC-SMG GSM~~ TS ~~03~~23.002

[3] ~~ETSI TC-SMG GSM~~ TS ~~03~~23.060

[4] ETSI TC-SMG GSM 11.14

[5] ETSI TC-SMG GSM 30.01

[6] ~~ETSI TC-SMG GSM~~ TS 23.001.

[7] TG.3x6.

[8] UMTSYY.01, UE-UTRAN Radio Interface Protocol Architecture – Stage 2

[9] UMTSYY.03, Description of UE states and Procedures in Connected Mode

3 Definitions and abbreviations

3.1 Definitions

Editors note : Reference to Definition document required.

For the purposes of the present document, the [following] terms and definitions [given in ... and the following] apply.

<defined term>: <definition>.

example: text used to clarify abstract rules by applying them literally.

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

<ACRONYM> <Explanation>

4 UMTS Concepts

Section 8 contains concepts that are considered as stable within SMG12 and no further input is expected but it should also be noted that consensus could not be reached on their use within UMTS.

4.1 Reduction of UMTS signalling

4.1.1 GLR Concept

The benefits of the Gateway Location Register (GLR) are:

- reduction in signalling traffic between networks.
- potential enhancements to mobile terminated call handling

4.1.1.1 Overview of the GLR Concept

The GLR is a node between the VLR and the HLR, which may be used to optimise the handling of subscriber location data across network boundaries.

In Figure 1, the GLR interacts with HLRa and VLRb for roamers on Network B. The GLR is part of the roaming subscriber's Home Environment. When a subscriber to HLRa is roaming on Network B the GLR plays the role of an HLR towards VLRb and the role of a VLR towards HLRa. The GLR handles any location change between different VLR service areas in the visited network without involving HLRa.

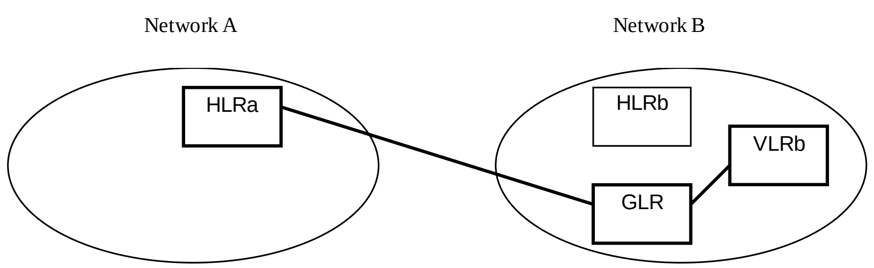


Figure 1: GLR Overview

The sequence of events when the subscriber roams to network B is as follows:

- VLRb sends the registration message to HLRa via the GLR, (i.e. HLRa stores the GLR's SCCP address and the GLR stores VLRb's SCCP address).
- HLRa returns the subscriber profile data
- The subscriber profile is stored in the GLR and VLRb

As the roaming subscriber moves between VLRs in network B, then the GLR is updated, but no message is sent to HLRa, therefore the number of messages between Network A and Network B is reduced. The reduction in signalling traffic is a significant benefit when the two networks are far apart, e.g. between Europe and Japan.

4.1.1.2 Applications of the GLR

In addition to reducing the amount of mobility related signalling between networks, the GLR's function might also be extended to other aspects. These include the following:

- Enhancements for mobile terminated call handling
- Support for the Virtual Home Environment of a roaming subscriber
- Reduction of CAMEL signalling traffic between the visited and home network
- Hiding local variations in signalling between networks
- Further study is needed on these issues

4.1.2 Super-Charger

The signalling load associated with subscriber roaming can be high when either the MSC/VLR areas are small or the subscriber travels significantly. The Super-Charger concept aims to optimise signalling associated with subscriber data management by retaining subscription data in previously visited VLRs, where possible.

The benefits of the Super-Charger concept are:

- Reduction of signalling traffic for subscribers located in the home PLMN,
- Reduction of signalling traffic between the visited PLMN and the home PLMN,
- No new network nodes are required,
- Applicable to a wide range of protocol used for the transfer of data.

4.1.2.1 Overview of the Super-Charger Concept

The concept of the Super-Charged network is described with examples from GSM mobility management. However, Super-Charger can be applied to other scenarios and protocols. This is a further study.

Super-Charger retains subscriber data stored in VLRs after the subscriber has moved to a location area served by a different VLR. The HLR performs the insertion of subscriber data to the VLR serving the location area to which the subscriber has roamed. The subscriber data stored at previously visited VLRs shall not be maintained while the subscriber is located in a location area serviced by a different VLR.

When the subscriber moves to a location area served by a VLR that has retained the subscriber's subscription data, the VLR shall indicate to the HLR whether subscriber data is required. If the VLR indicates that subscription data is not required but the user's subscription data has changed the HLR shall send the new subscription data to the VLR. Figure ~~x~~2 shows an example message flow in a Super-Charged network.

To ensure data consistency for super-charged VLRs a sequence numbering method can be used. A sequence number is added to the subscriber data record. This sequence number is incremented whenever the subscriber data record is changed for any reason. The sequence number is sent to the VLR in ISD. For non-super-charged VLRs this can be ignored. For super-charged VLRs it is stored and returned to the HLR in subsequent UpdateLocation messages. The HLR can then compare this sequence number with the value currently stored in the HLR to determine if the cached data is still valid.

With the Super-Charger activated subscriber information is no longer deleted from the VLR database when a mobile station moves from the location area served by the VLR. This results in the continuous growth of the VLR database size. Consequently, a new VLR data management system is required so that the VLR can handle newly arrived mobile stations. Two options for subscriber data management systems are:

- subscriber data for subscribers that are not currently served by the VLR shall be deleted periodically using a VLR audit system and/or,
- subscriber data for subscribers that are not currently served by the VLR shall be deleted dynamically to make room for the newly arrived subscribers.

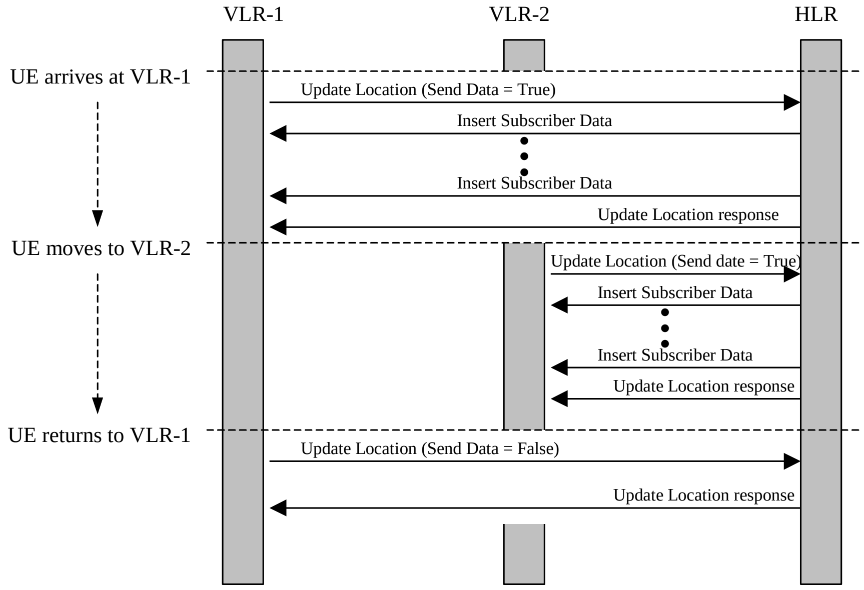


Figure 2: Example message flow in a Super-Charged network.

4.1.3 Turbo Charger

The signalling load associated with subscriber roaming can be high when either the location areas are small or the subscriber travels significantly. The Turbo-Charger concept aims to optimise signalling associated with subscriber data management by assigning one MSC/VLR to perform the Call Control and Mobility Management functions while the subscriber remain attached or until signalling routes require further optimisation.

The benefits of the Turbo-Charger concept are:

- the substantial reduction in signalling traffic for subscribers located in the home PLMN,
- the substantial reduction in signalling traffic between the visited PLMN and the home PLMN,
- no new network nodes are required,
- applicable to a wide range of protocol used for the transfer of data.

The disadvantages of the turbo-charger concept are:

- Connections are required from the access network to be fully meshed to all MSCs in the turbo-charger area.

4.1.3.1 Overview of the Turbo-Charger Concept

A Turbo-Charged network constitutes a network architecture designed to reduce mobility management costs and provide automatic load-sharing between MSC/VLRs.

The architectural philosophy is to equally divide the subscribers between the available MSC/VLRs, irrespective of their location. In the context of GSM, this could be achieved by placing a routing function (e.g. evolved STP) between the BSC and the pool of MSC/VLRs. The purpose of the routing function is to route A-interface messages to the MSC/VLR that is serving the mobile station. The solution requires the MS to store a discriminate that can be used to identify the serving MSC/VLR and for routing to be applied on this discriminate on the connection between the MSC/ VLR and access network. A TMSI partitioning scheme could be utilised. This scheme allocates a sub-set of the TMSI range to each MSC/VLR, Figure 3. The A-interface messages are then routed to the right MSC based on the TMSI. This could be done by a routing function external to the access network implying no access network modification (see figure 3). If a TMSI partitioning scheme is used then new SIM cards are not required.

The temporary identity used for paging (TMSI) must be unique within all the MSCs in the turbocharger area. This implies that there must be a mechanism to ensure that this requirement is met for turbocharged MSCs (e.g. TMSI partitioning).

Two mechanism to provide load-sharing are envisaged, random load-sharing and dynamic load-sharing.

Random load-sharing requires the routing function to randomly assign a MSC/VLR to serve a particular mobile station when it first comes in to the network. Regardless of where the mobile is the same MSC/VLR will always serve it provided the mobile remains in the area served by all the turbocharged MSC/VLRs linked by the routing function.

In large metropolitan areas where subscribers are served by multiple MSC/VLRs, some MSC/VLRs may be very busy while others are not fully utilised. Dynamic load-sharing requires the implementation of an intelligent router. Since the routing function routes all A-interface traffic, it can participate in load-sharing and balancing based on the current loading of each MSC however linkage between MSC load and the routing algorithm would be required.

In the case of a Turbo-Charged network where the network is sub-divided into large regions, further optimisation can be achieved by adding the Super-Charger functionality.

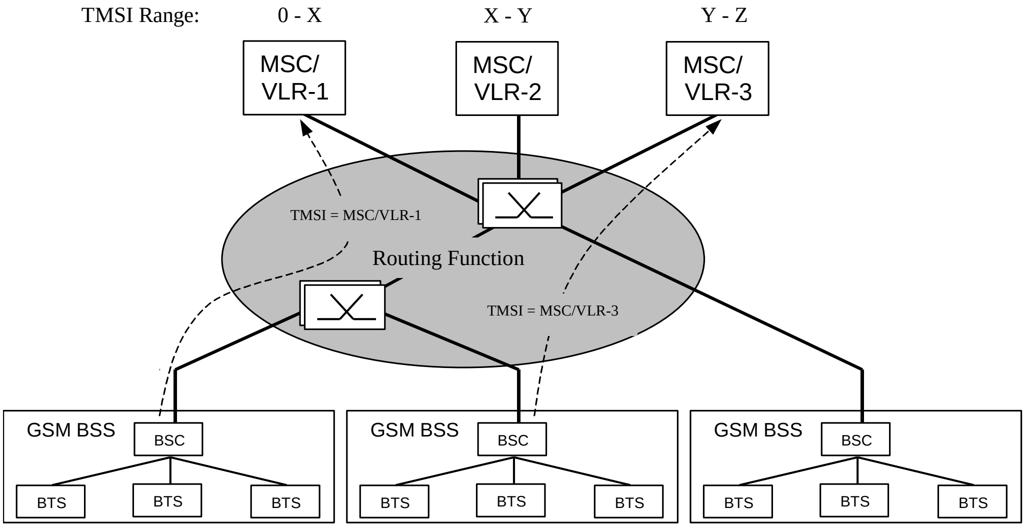


Figure 3: Example of GSM Turbo-Charger Network Architecture

In the context of UMTS, the routing function becomes a feature of the RNC.

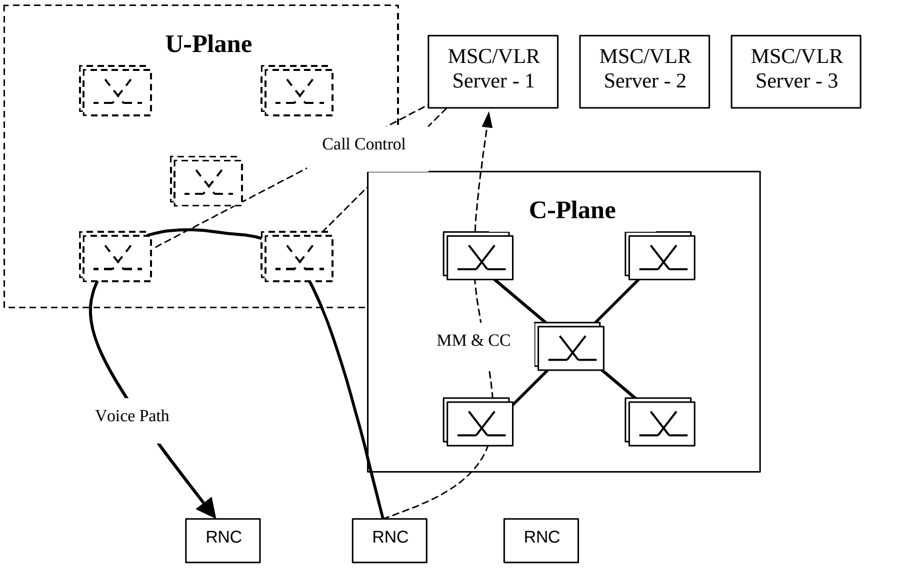


Figure 4: Example of UMTS Turbo-Charger Network Architecture

4.1.4 Relationship between GLR and TurboCharger

The GLR and TurboCharger are two independent schemes for reducing the amount of MAP traffic generated in UMTS networks.

- The GLR works by reducing traffic between PLMNs associated with Location Updates. This is achieved by "caching" the roaming subscriber's data in the visited network
- The TurboCharger works by eliminating the need to perform location updates. The same VLR can hold a subscriber's data for the duration of his attachment to the network.

A TurboCharged network requires that each MSC/VLR can physically connect to all RNCs. Therefore TurboCharging may be best suited to areas of the network characterised by dense geographic coverage. On the other hand, the GLR function is independent of the network density.

The network structure shows that the GLR and a TurboCharged area within the same PLMN are independent. In fact, it shows benefits from using the two techniques in the same network. The Turbo-Charger reduces the location registration signals between the MSC/VLR and GLR:

- There is no new update location signal between MSC/VLR and GLR if roamer moves inside of the Region A.
- There is no new update location signal between GLR and HLR if roamer moves between regions.

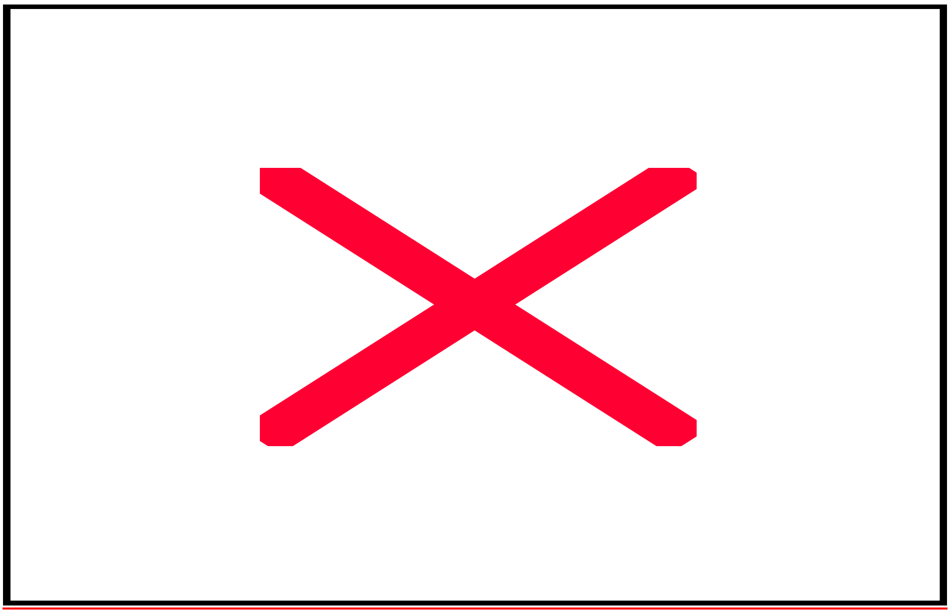


Figure 4bis. [editor's note: to be deleted]

5 Key issues

{Editors note: These key issues have arisen from the scenario work, it is agreed within SA 2 that the focus should be on solving these key issues, Once these issues have become relatively stable, they are moved to 23.121 or removed from this document}. Study of these items is ongoing.

5.1 Core network transport

- L1 and L2 technologies
- Signalling protocols
- How to use ATM?
- Nx64k transport

5.2 Core network layer 3

5.2.1 Common Communication Channel

A common communication channel (name to be defined) provides nodes of the Core Network the ability to reach every RNC of the UTRAN. This communication channel can be used for application like SMS cell broadcast or location services (LCS).

This communication mechanism would use e.g. an IP routing functionality of the 3G-SGSN. The according protocol stack is outlined in figure 5.

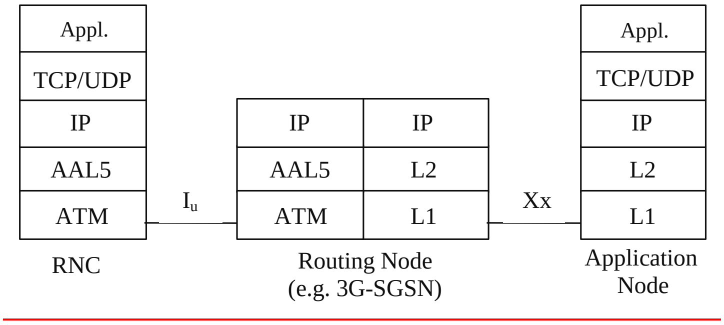


Figure 5: Protocol Stack of the Common Communication Channel

The placeholder Xx should be replaced by the according reference points of the applications e.g. Bc for cell broadcast.

The following issues until now are identified and have to be solved:

1. IP Routing functionality in the 3G-SGSN,
2. An appropriated layer 3 protocol has to be chosen (TCP or UDP) per application,
3. Addressing of the Application and Application node by the RNC(s),
4. Addressing (dynamic or static) of the application (e.g. CBC) on the RNC(s).

- L3 technologies
- GTP vs. IP-in-IP tunneling

In UMTS/GPRS, it should be possible for operators to use different packet switching protocol (e.g. ATM-SVC) under single GTP standard.

Between GSNs GTP uses UDP/IP (or TCP/IP) for addressing regardless whether IP routing or ATM-SVC switching is used. The use of ATM-SVC will not impact on GTP standardisation

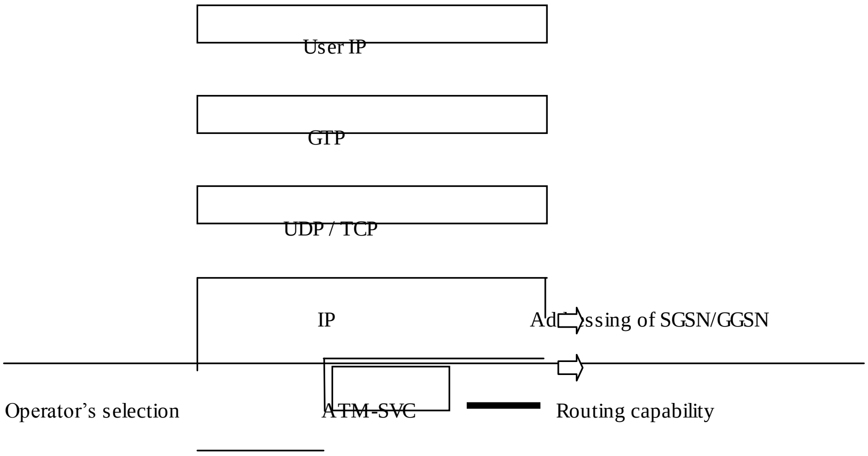


Figure 6

5.3 Benefits of the Gs interface applied to UMTS

The Gs interface defined within GSM/GPRS provides a number of benefits to a GSM/GPRS operator [03.60]. These include: combined attach/detach procedures, combined location/routing area updates, paging of CS connection via the SGSN, identification procedures, MM information procedures. The main aims of these include saving of GSM/GPRS radio resources, harmonised security procedures and reduction of MS battery consumption.

As GSM operators roll out GPRS and as the numbers of mobiles increase the benefits of the Gs interface to the network operator will increase as the percentage of GPRS enabled mobiles grows. GSM/GPRS operators with mature networks will also be looking to roll out UMTS using evolved CN infrastructure, they will also be looking to apply the benefits of the Gs interface reaped for GSM to UMTS. Many of the capabilities of the Gs interface will be applicable to UMTS (such as combined updates, combined attach and MS/Ue information procedures), this will save on radio resource usage. The presence of the MSC-GSN interface will also offer the opportunity for developments to ease seamless service support between CS and PS platforms (such as SoLSA and Camel).

In the future, network operators who have incorporated Gs functionality into their networks will be looking to connect UTRAN to their GSM/GPRS Core Networks with minimal changes (excepting those for service development, network and radio optimisation, network evolution and flexibility), thus the Gs interface should be maintained and enhanced for UMTS.

The Gs interface also offers opportunities for suppliers and operators regarding integrated MSC/GSN products (which may support internal proprietary Gs functionality as well as standardised MSC-GSN functionality). Operator's networks which have separated MSC/GSN nodes will be able to add integrated nodes into their GSM/GPRS/UMTS networks (and vice versa), depending upon the MM solutions developed for UMTS this could enable combined updates to be performed between (Gs supporting) integrated and separated nodes. If the Gs interface is not present operators will not be able to optimise resource between (integrated or separated) nodes.

5.3.1 Periodic updates

5.3.1.1 Why do we have Periodic updates

Periodic updates are within the network to increase the efficiency of the CN while also increasing the quality of service perceived by calling parties to mobiles. The periodic timer is set within the CN node to a figure which enables absent mobiles to have their (VLR based) information removed after the timer expires. People calling mobiles which are registered as 'detached' (either implicitly or via periodic expiry) will receive faster treatment of the call in the CFNRc case or 'Not been possible to connect your call' RANN case as the mobile is not paged by the network.

5.3.1.2 Support of periodic updates in UMTS

One of the current proposals for SRNS relocation [1, incl.: section 9.3.4, 2] propose that when in CMM connected mode (PMM idle) or PMM connected (CMM idle) the relevant location/routing updates to the (idle) CN are performed while in RRC connected mode.

For periodic updates the UE may be RRC connected (know to the UTRAN as 'active') when the (UE based) periodic timer is due to expire, the (idle) CN node will also have a timer about to expire and be ready to detach the UE.

If the methodology of [1, Section 9.3.4] is followed a location update will be performed within the same RRC connection to the (MM idle state) CN node to re-set the periodic timer.

5.3.1.3 Impact upon UMTS

The impact upon UMTS of this is that the UTRAN, UE and one CN node have an active session ('xMM connected) in place with accurate knowledge of the (periodic) attached/detached status of the UE. It is a waste of (valuable) radio resource for the UE to perform a LA/RA update purely to reset the periodic timer in the (idle) CN node: this also contradicts working assumption [1, section 11].

As UMTS is envisaged as a mass market system supporting very large numbers of mobiles within the network, many of these could potentially have very long (i.e. all day) duration (but low packet volume) Packet sessions (as per GPRS). It is folly to consider additionally loading the radio resource to update the (periodic) detach status of the mobile on the CN side of the radio interface when elements on the CN side of the radio interface already know the status of the mobile.

5.4 Authentication

5.5 Management of ciphering keys

5.5.1 Cipher Mode Control – 2MM concept

The assumptions in this section is based upon the assumption that ciphering in performed between UE and RNC.

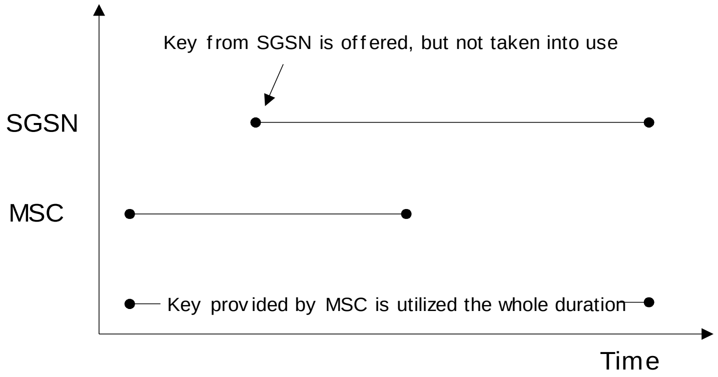
It is assumed that in UMTS the ciphering key and the allowed ciphering algorithms are supplied by CN domains to the UTRAN usually in the beginning of the connection. Receipt of the ciphering command message at the UTRAN will cause the generation of a radio interface ciphering command message and, if applicable, invoke the encryption device and start data stream ciphering. The CN domain is noted if the ciphering is executed successfully in the radio interface and the selected ciphering algorithm.

When new connection is established from other CN domain, which is not having any connection to the UE, the new CN domain also supplies the ciphering key and the ciphering algorithms allowed to use to UTRAN in the beginning of the connection. This is due to the fact CN domains are independent from each other.

5.5.1.1 One ciphering key used in UTRAN

If it is assumed that only one ciphering key and one ciphering algorithm are used for all connections, this leads to a situation, in which there are two ciphering keys supplied from CN domains and only one of them is used.

To handle this situation, UTRAN must select either one of the ciphering keys. If there are no differences between the ciphering requirements[1] requested by two CN domains then, e.g., the first ciphering key and the algorithm is maintained (see Figure ~~3~~ 7).



~~Figure 3~~Figure 7. One ciphering key use in the UTRAN

As a result of the selection of the ciphering key between two different CN domains (if both CN domains have active connection(s) to the UE) either one or both of the CN domains do not know the present ciphering key used for the connection(s). Only UTRAN and UE know the present ciphering key used.

Further, if the case described in figure 1 is still considered and if after the MSC connection is released, but before SGSN connection is released, a new connection from MSC is established, the MSC may initiate a new authentication resulting in a new MSC ciphering key supplied to UTRAN. In this case, the UTRAN may follow the same key selection approach as it used previously, i.e., the first ciphering key is maintained[2].

5.5.1.2 Multiple ciphering keys used in UTRAN

It may be required to use more than one ciphering key for different radio access bearer, e.g., user plane bearers associated to one CN domain are ciphered by the ciphering key supplied by the associated CN domain. However, in the control plane only one ciphering key is used and therefore in the control plane there must be co-ordination between ciphering keys supplied by CN domains.

The co-ordination in the control plane is similar to what is presented for one ciphering key used in UTRAN option (ch. 2.1). In the control plane, UTRAN must select either one of the ciphering keys supplied from CN domains if both CN domains are active. The change of the used ciphering key in the control plane during active RRC connection is for further study.

5.5.1.3 Serving RNC relocation and ciphering

In GSM, when inter-BSC handover is performed, MSC sends the ciphering key and allowed algorithms to the target BSC in the BSSMAP HANDOVER REQUEST message. In GPRS, because the SGSN performs the ciphering, the inter-BSC handover does not cause any need for the ciphering key management.

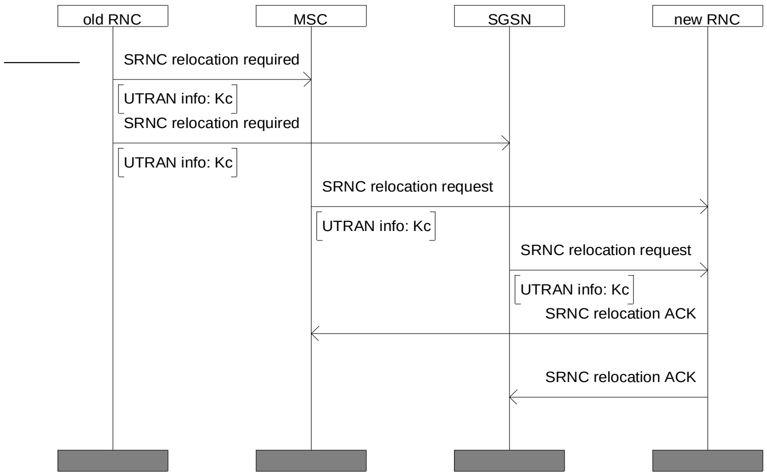
For UMTS, the GSM approach is not applicable on the serving RNC (SRNC) relocation, because CN domains do not necessary know the present ciphering key(s) used as it is described in the chapter 2.

It is recommended that the ciphering key(s) or a relevant information indicating used ciphering key(s) is transferred in the transparent UTRAN information field from the source RNC to the target RNC in the RANAP SRNC RELOCATION

[1] E.g. a requirement for more efficient ciphering algorithm that is currently used for the connection(s).

[2] The change of used ciphering key during an active RRC connection is considered as a further study item.

REQUIRED and RANAP SRNC RELOCATION REQUEST messages (see Figure ~~4~~8.). In this way the present ciphering key(s) is transferred to the target RNC.



~~Figure 4~~Figure 8. The ciphering key transfer in SRNC relocation procedure (one ciphering key)

5.5.2 UMTS-GSM handover

In the handover from UMTS to GSM, the ciphering key cannot be transferred transparently like it is proposed for UMTS. The CN has to build the BSSMAP HO REQUEST message, having the ciphering key from the MSC. 2G-SGSN receives its ciphering key from the old 3G-SGSN via Gn-interface as it is done in GPRS.

If the ciphering keys used in UMTS are different compared to GSM, e.g., the ciphering key length is different, both MSC and SGSN ciphering keys must be changed in UMTS-GSM handover. This type of interoperation is left for further study in this paper.

5.5.3 Interworking with 2g-MSC

In GSM, the A-interface BSSMAP [2] supports a transparent field in the BSSMAP HO REQUIRED and HO REQUEST messages, which allows to utilise the proposed solution also for GSM CN connected to the UTRAN.

5.6 Mobile IP in UMTS

5.6.1 Mobile IP

A single generic mobility handling mechanism that allows roaming between all types of access networks would allow the user to conveniently move between fixed and mobile networks, between public and private as well as between PLMN's with different access technologies. The ongoing work in IETF Mobile IP working group [MIP WG] is targeted towards such a mechanism[3]. Thus it is important to offer Mobile IP also to UMTS users and UMTS must be developed to support Mobile IP. Mobility within the UMTS CN could also be handled by Mobile IP. This would allow transparency to networks external to the UMTS PLMN. Potentially, this would allow cost savings for operators and a broadening of the market for manufacturers.

[3] Note that in this text, Mobile IP is used in a wide sense. It refers to [RFC2002] **and** ongoing and future work in the IETF Mobile IP Working Group [MIP WG].

It is important to understand the different driving forces:

- Mobile IP as an overlay to the UMTS-GPRS would make it possible to offer easy roaming between different types of networks
- An integration of Mobile IP within the UMTS CN would additionally allow the operators to use standard IP technology to a larger extent and thus lower the cost for deployment and maintenance of networks.

Operators shall have the possibility to offer Mobile IP to end customers for R99. A flexible approach should be taken in order to extend the use of Mobile IP to handle mobility within the UMTS CN. UMTS standards should be aligned to when new Mobile IP functionality, that is needed for the different steps, will come out on the market. As not all operators will introduce Mobile IP at the same time, compatibility with GPRS based PLMN's is needed. Such a flexible, yet backward compatible, approach is outlined below.

The concept of surrogate registration [TEP] allows MS's without Mobile IP to benefit from Mobile IP infrastructure by letting the network perform the registration with the HA on behalf of the MS. However, this issue needs further investigation.

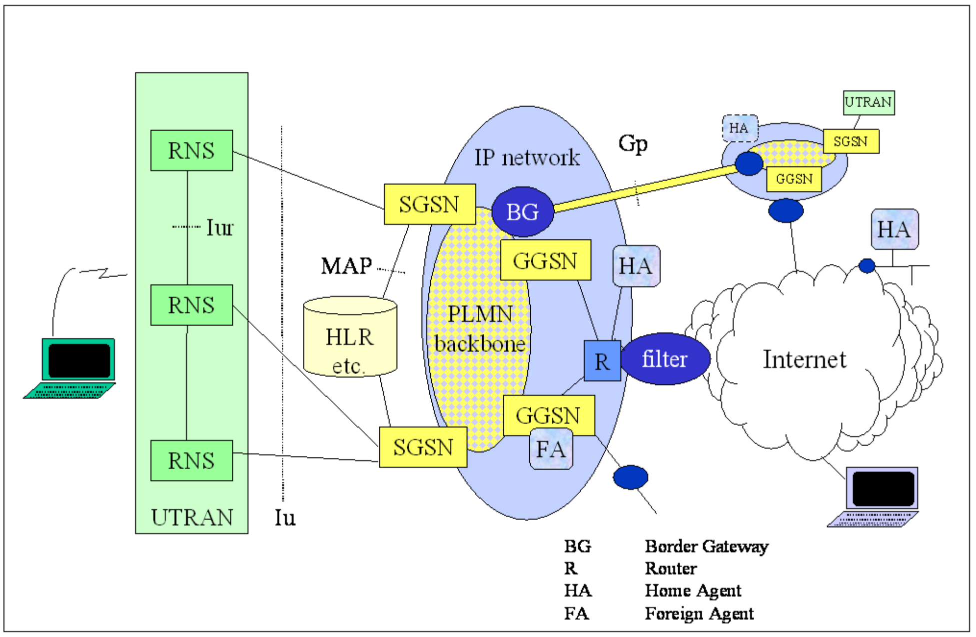
5.6.2 A staged introduction of Mobile IP in the UMTS CN

Three steps, which are discussed more in detail further down, have been identified. Briefly, these are:

1. 1. Step 1 represents a minimum configuration for an operator, who wishes to offer the mobile IP service. The current GPRS structure is kept and handles the mobility within the PLMN, while MIP allows user to roam between other systems, such as LAN's, and UMTS without loosing an ongoing session, e.g. TCP.
2. 2.The SGSN and GGSN can be co-located without any alterations of the interfaces. However, to obtain more efficient routing, the MS could change GGSN/FA, i.e. PDP context and care-of address after an inter SGSN handover if it is not transferring data. MS's which are transferring data during the inter SGSN handover could perform the streamlining after the data transfer is completed, using the old GGSN as anchor during the completion of the data transfer.
3. 3. The third step is to let MIP handle also handover during ongoing data transfer. The Gn interface is here only needed for handling roaming customers without support for MIP.

5.6.2.1 Step 1 – Offering Mobile IP service

Mobile IP has the benefit of being access system independent, which allows users to roam from one environment to another, between fixed and mobile, between public and private as well as between different public systems. Assuming a minimal impact on the GPRS standard and on networks whose operators do not wish to support MIP, leads to the following requirements:



~~Figure 5~~Figure 9. Core network architecture with GPRS MM in and between GPRS PLMN's and Mobile IP MM between different types of systems and optionally between GPRS PLMN's.

- The MS must be able to find a FA, preferably the nearest one. The underlying assumption is that FA's are located at GGSN's and that not all GGSN's may have FA's. One FA in a PLMN is sufficient for offering MIP service, however for capacity and efficiency reasons, more than one may be desired. This means that the MS must request a PDP context to be set up with a GGSN that offers FA functionality.
- While setting up the PDP context, the MS must be informed about network parameters of the FA, e.g. care-of address.
- Furthermore, the interaction between the GGSN and the FA needs to be studied more in detail. With the assumption that FA care-of addresses are used, the FA needs to detunnel incoming packets and, together with the GGSN, map the home address of the MS to a PDP context.

Roaming can be handled either via the Gp interface or via Mobile IP. This is described in the section on roaming further down. It is assumed that the MS keeps the same care-of address as long as the PDP context is activated.

A typical network is shown in Figure ~~5~~9. The detailed solutions of this step are to be worked out in the Mobile IP technical report.

5.6.2.2 Step 2 – Intermediate GPRS-Mobile IP system

One way to implement a GPRS backbone is to co-locate the SGSN and GGSN, as depicted in Figure ~~6~~ 10 . This might be favourable for operators with a strong interest in utilising standard IP (IETF) networks as far as possible and does not require any changes in the current GPRS protocol architecture.

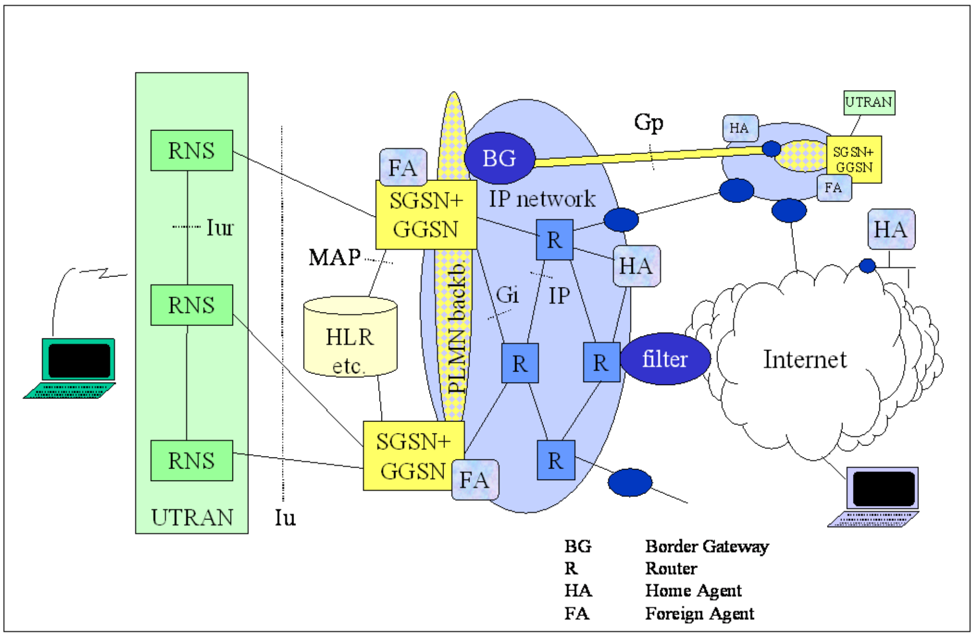
In step 1, the assumption was that the MS stays with the same care-of address, during a session, i.e. as long as a PDP context is activated. A very mobile MS, might perform several inter SGSN HO's during a long session which may cause inefficient routing. As an initial improvement, a streamlining procedure, with a temporary anchoring point in the GGSN, could be introduced:

If the MS is not transferring data while moving from one SGSN to another, a new PDP context could be setup between the new SGSN and its associated GGSN at the handover. The MS will get a new care-of address. The procedure for informing the MS that it has arrived to a new network has to be defined.

If the MS is transferring data, e.g. being involved in a TCP session, the MS would move from the old SGSN to the new one while keeping the PDP Context in the old (anchor) GGSN for the duration of the data transfer. Once the data transfer is terminated, the PDP Context can be moved to the GGSN associated with the new SGSN and a new care-of address can be obtained.

The buffer and forward mechanism, which already exists between the SGSN's for preventing data loss at inter SGSN HO's, will, with this procedure, be reused as it is. This procedure also has some advantage regarding the handling of firewalls, which are assumed to be attached to the GGSN's. Today, there is no standard for changing firewall during e.g. a TCP session.

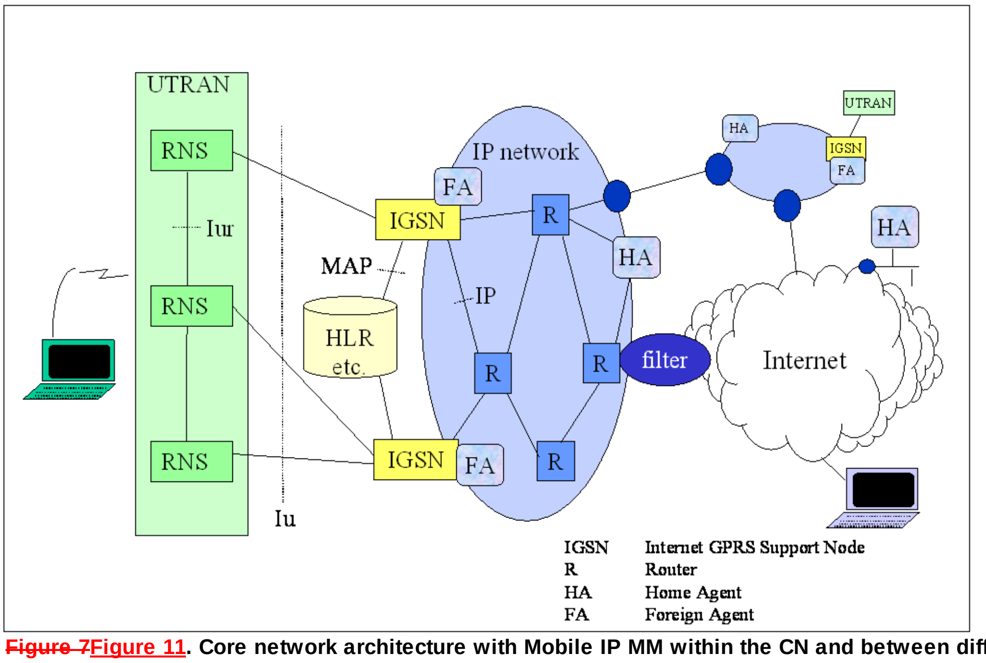
As in the previous step, the GPRS interfaces (Gn and Gp) need to be deployed for roaming customers, since there might be networks which not yet supports Mobile IP. Roaming between PLMN's can be handled either with Mobile IP or with GPRS.



~~Figure 6~~Figure 10. Core network architecture where GPRS MM handles active mobiles and Mobile IP streamlining at inter SGSN handover. The SGSN and GGSN are here co-located.

5.6.2.3 Step 3 – Using Mobile IP for Intra System Mobility

The third and last step is to let Mobile IP handle all intra system mobility, including all handovers between GGSN's or IGSN's. This is depicted in Figure ~~7~~11, where the IGSN represents an integrated SGSN/GGSN. The Gn and Gp interfaces may optionally be kept to handle roaming customers, whose terminals do not support MIP and the operator's own customers roaming to networks without MIP functionality. The main difference compared to the previous step is that lossless handovers between IGSN's must be handled. This architecture is investigated by the Mobile IP ad hoc group in a feasibility study.



~~Figure 7~~Figure 11. **Core network architecture with Mobile IP MM within the CN and between different types of systems and between GPRS PLMN's.**

5.6.3 Roaming

Depending on the capabilities of a visited network, two roaming schemes can be identified; GPRS roaming and MIP roaming. With GPRS roaming, we mean roaming via the Gp interface and the use of a GGSN in the home network, which is necessary when the visited network does not offer any FA's. In those cases where the visited network offers a FA, either a GGSN/FA in the visited or in the home network can be utilised. Networks, which use Mobile IP for all its own customers can provide GPRS roaming to visiting users by deploying the Gn and Gp interfaces.

5.6.4 Mobile IP and UMTS terminals

The mobile equipment needs to be enhanced with MIP software. For compatibility with other systems, it is of great importance that standard IETF Mobile IP and not a special UMTS version is used. Although it should be kept to a minimum, any interaction between the IP layer and the "UMTS layer" needs to be identified and defined. To avoid future updates of the mobile equipment, which is supposed to support Mo bile IP, it should be considered to include the UMTS specific functionality, needed to support Mobile IP in all three steps in the MS at once.

Surrogate Registrations

The concept of surrogate registration has a potential use in supporting non Mobile IP aware terminals using a Mobile IP based infrastructure. Instead of the MS performing registration with the Home Agent according to [RFC 2002], the FA could surrogate the mobile node in performing Mobile IP registrations with the Home Agent. One solution is proposed in [TEP] (Tunnel Establishment Protocol). However, surrogate registration may cause IP level authentication to be dependent on UMTS authentication and hence increase the dependence of Mobile IP on the access technology. Further study is required on this topic.

5.7 Iu reference point

5.7.1 General

As a first step, UMTS will be based on the GSM/GPRS network, i.e. one circuit switched and one packet oriented domain. Due to the differences of the domains, the Iu reference point will be realised by two Iu instances, one for each domain. This enables each domain to develop according to their specific characteristics. At the same time, an aligned view of the Iu reference point should be achieved where this is deemed suitable

5.7.2 Control structure for the Iu reference point.

- A multi-vendor interface shall be defined at the Iu reference point (Iu interface). The interface embodies a protocol suite allowing different protocol stacks towards the PSTN/ISDN domain and the IP domain.
- Over the Iu interface, user information to one UE is carried in one or several logical user flows, controlled by a signalling protocol (RANAP). Additionally some control elements (potentially relevant for only one domain) may be carried inband in the user flows.
- A common syntax for RANAP messages for both the IP and the PSTN/ISDN domain is the target as long as the functionality of either domain is not compromised.
- A guideline for defining the control procedures over the Iu reference point is to reuse, to the extent possible, control procedures defined in BSSMAP and BSSGP/GTP. The use of BSSMAP and BSSGP/GTP as the base when defining the control procedures over Iu does not preclude new control procedures to be introduced over Iu reference point.
- For each domain the protocol stack used by RANAP may be based on one of SS7, TCP/IP or a combination (e.g. SCCP on TCP/IP or UDP/IP). The protocol stack used by RANAP may be different for the PSTN/ISDN domain and the IP domain.
- The protocol stack used by the user data transport over Iu may be different from the protocol stack used by RANAP. Furthermore the user plane protocol stack may be different for the two domains.

5.7.3 Iu reference point – User plane towards IP domain

- Any problems within the UTRAN which cause loss of data addressed to a UE shall be indicated to the 3G-SGSN to maintain the conformance of the data volume counted by the 3G-SGSN with the successfully transferred data volume. It is FFS whether this mechanism provides a degree of conformance required for volume dependent charging.

5.8 Dualmode operation (GSM/UMTS)

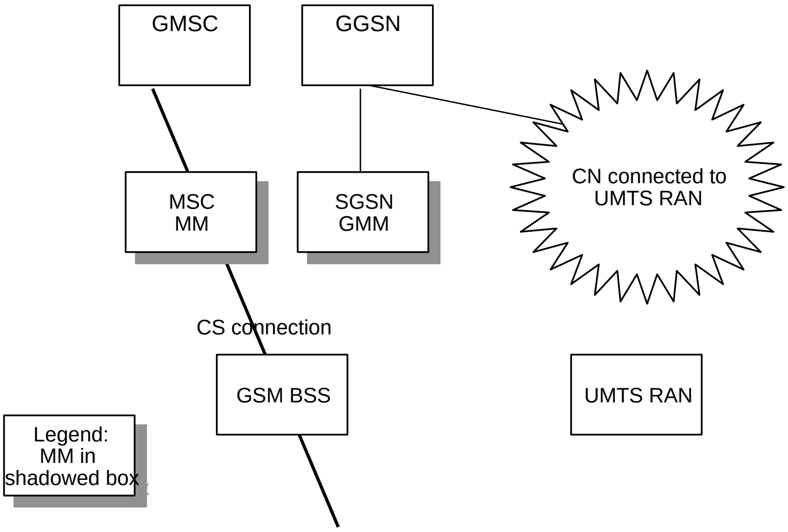
5.8.1 Will dualmode terminals also support GPRS?

5.8.1.1 Handovers between GSM/GPRS class A and UMTS terminals

In the following some problems and suggestions to solve the problems are made concerning the case where UMTS must support handovers from GSM to UMTS and/or UMTS to GSM for mobile stations with CS and PS service capability (GPRS class A).

5.8.1.2 Handover from GSM to UMTS

This type of handover could be needed, e.g., due to traffic reasons in a congested GSM network. In GSM the control for CS connection remains in the MSC from which the call was originated. This is called anchoring. Figure ~~8~~ 12 illustrates the situation before the HO into UMTS (i.e., to UMTS UTRAN).



~~Figure 8~~Figure 12. Before HO to UMTS from GSM. The PS services are provided from SGSN in GSM.

In order to have access to PS services after the HO, the MS has to perform the necessary update, obviously. The reason for this is that there are no means to change SGSN in GSM without doing so. However, as there is an active connection from GSM MSC, no updating can be done for CS services (i.e., to MSC connected to UMTS UTRAN) until the call has ended, i.e., the control and MM for CS remains in GSM. As a result, only the PS MM can be activated in UMTS and thus the MM is split into two due to the HO.

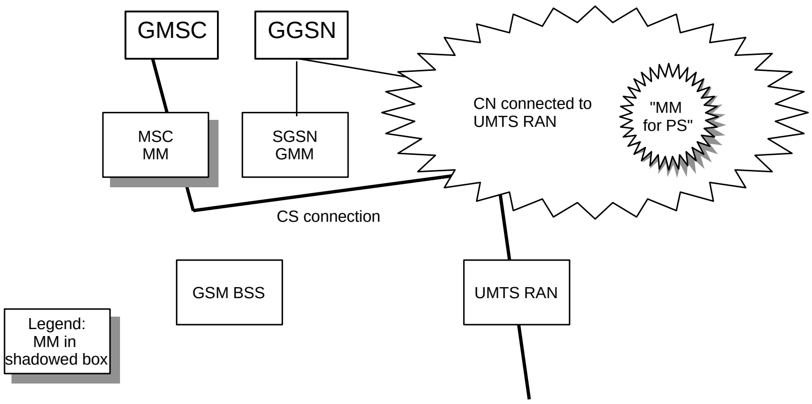
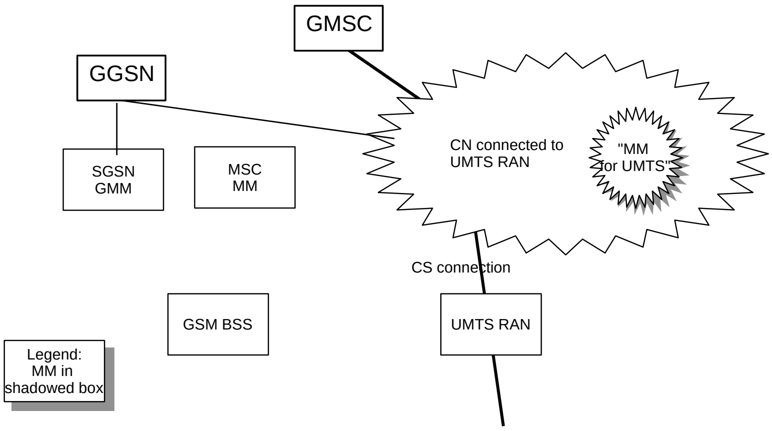


Figure ~~9~~ 13 After HO to UMTS from GSM.

The PS services can only be accessed from UMTS CN. To avoid severe limitation on accessing the PS service during the length of the CS connection, "MM for PS" must be setup into UMTS CN. To support that, the MM in UMTS CN has to be able to be split into two like MM in GSM. Moreover, to support PS access the UTRAN needs to perform co-ordination similar to the one required for the core network architecture with two edge nodes (e.g. scenario 2).

5.8.1.3 Handover from UMTS to GSM

Figure ~~10~~ 14 illustrates the situation before the HO; anchoring is assumed in UMTS CN. This type of handover could be needed, e.g., due to limited coverage of UMTS.



~~Figure 10~~ Figure 14 Before HO from UMTS to GSM.

This type of handovers are seen as important especially in the first stages of UMTS due to limited coverage. Without these, the end user perception may be seriously affected.

Again, to have access to the PS services after the HO, an appropriate update is needed and also no updating can be done for CS. As a result, MM instance only for PS can now be activated in GSM as long as the call lasts and as a result, the MM in UMTS is split into two due to the HO..

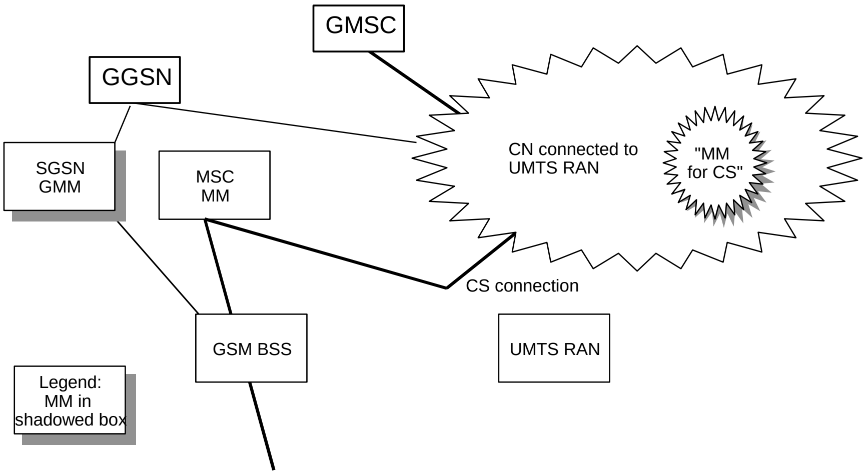


Figure ~~11~~ 15 After HO from UMTS to GSM.

To have access for PS service in GSM, the "PS part of MM in UMTS" has to be transferred to GSM.

5.8.1.4 Suggestions

From the discussion above one can suggest that to support handovers between UMTS and GSM for class A type of mobiles:

4. UMTS MM must support some distinction between CS and PS services in the registration related procedures. An example is a dedicated update/cancel only to PS services in UMTS. This is likely to affect to the states of UMTS MM sublayer in MS and CN (independent of the selected MM solution)

5. The MS has to be capable of handling the GSM – UMTS dualism

6. The UTRAN has to support the operation. Required functions bear resemblance to the architecture where the core network has two edge nodes (MSC, SGSN).

Some of these problems may be alleviated if the UMTS core network node provides also GSM functionality (A and Gb) and there is no need to change the UMTS core network node during the handover. This is for further study.

Requirements due to handover for dualmode "UMTS class A" – GPRS class B terminal are ffs.

- Handovers between GSM and UMTS for N-ISDN and packet oriented services (e.g. IP)
- Idle mode operation of dual mode terminals (e.g. cells in same or different location areas)

5.9 Anchor concept

UMTS Mobility Management (UMM) for release 99 shall use packet anchoring at the GGSN, providing this meets the QoS requirements, including those for real time services.

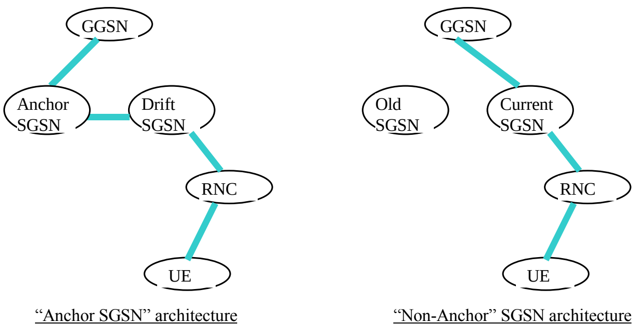
Disassociation of SRNS relocation and PS session transfer should be evaluated for release 99

5.9.1 Introduction to the concept of anchoring communications in GPRS

GPRS is being developed to include Quality of Service, this includes real time aspects. At present within GSM/GPRS the Core Network part of inter SGSN RA update procedure- is used to maintain communications within the network for a change of SGSN. GPRS will need development to support real time QoS requirements, the current mechanisms for changing the current SGSN (inter SGSN RA update) may also need developments to maintain the QoS requirements.

For UMTS the notion of Serving and Drift RNC provides a no loss of data at Hand-over inside a UTRAN as long as SRNS relocation (or a UMTS<=>GSM handover) is not performed (use of RLC between SRNC and UE in case of non-real time packet data, and use of soft handover in case of real time). The SRNC could be considered as an "anchor" point for the UTRAN. Therefore only the case of SGSN change induced by SRNS relocation has to be considered.

Within the UMTS CN two proposals have been made to satisfy the QoS requirements, the anchor SGSN concept and the non-anchor SGSN concept, both are illustrated in Figure ~~12~~ 16 and are discussed in the following sections.



~~Figure 12~~Figure 16: The "anchor SGSN" and the "non-anchor" SGSN architectures

5.9.2 The Anchor SGSN concept

This section proposes that the current technique for anchoring communications within the MSC is considered for application to the QoS based GPRS communications (i.e. between SGSNs). This technique is termed 'the Anchor SGSN concept' and is used to maintain the communications between the GGSN and the UE , with the SGSN(old) making a bearer link to the SGSN(new).

It should be noted that this concept may be applicable for UMTS as well as GPRS.

5.9.2.1 Requirements for the anchor SGSN

The requirements for the support of the SGSN anchor concept are discussed below

GPRS: With added QoS

To date GPRS has used a number of different QoS Criteria, however the GPRS (and UMTS) community have been looking at enhancing this to enable better support for real-time type features. The current Core Network GPRS inter SGSN RA update (SGSN change) relies upon the Old SGSN to suspend and buffer packet transmission, the new SGSN to interact with the GGSN/HLR to maintain the active session. The new SGSN then re-commences transmission and buffered packets (from the old SGSN) are passed to the mobile. The impact of this is potential breaks in transmission which would not satisfy Real-time/QoS requirements.

The Core Network part of GPRS Cell re-selection: convergence with (inter MSC) handover?

As GPRS adopts real time QoS, developments will be needed within the routing elements (GSN) to cater for the real-time nature of the packet communications. One upshot of this is within the QoS based environment the resource reservation paradigm moves towards a 'circuit switched' one (with resources 'reserved' for the QoS stream). With this in mind the support of the CN part of inter SGSN RA update in a QoS environment could become closer to a 'circuit switched' handover where the old and new paths are 'connected/bridged' during the actual handover. For UMTS the SRNS relocation within a QoS based GPRS network may require developments between SGSN to enable the paths to be connected at an inter SGSN level, rather than the current method of using the GGSN. Effectively the current GPRS inter SGSN RA update mechanism uses the GGSN as the anchor point.

To maintain the QoS requirements during a change of SGSN an SGSN based anchor point (similar to the current VMSC based anchor in GSM CS) could be applied. Following the successful SGSN change it may be possible to optimise the packet routing between the GGSN and new SGSN, this requires further study.

SGSN based Anchor

The adoption of an SGSN based anchor could ease some of the problems highlighted within UMM~~}~~ where the MM becomes split between GSM/GPRS and UMTS when a handover between the two radio mechanisms occurs. At present the (GPRS) MM location follows the Packet Switched serving node (SGSN) as it moves within and between the networks, whereas the Circuit switched (CS) MM remains within the anchor MSC. Further study should be made to see if the concept of anchoring of all services within the 'initial' network (network where communications were initiated) will ease the 'split MM' problem.

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MM enhancements

Within the Circuit switched world the MM information is retained at the old MSC following an inter MSC handover and no location update is performed until the CS session (call) has been terminated. The adoption of a similar mechanism for Packet switched Services could ease the GSM-UMTS MM problem. If a CS session is in place the location update/routing area update would be constrained until the CS session is terminated (from the UTRAN perspective any PS packets would be routed over the common RRC session with no need for paging (in the now 'new' UTRAN RA/LA)).

5.9.2.2 Developments of GSM/GPRS for the SGSN based anchor

To enable the SGSN anchor concept to be supported the following developments will be needed to the contemporary GSM/GPRS network: these should be linked in to the overall UMTS developments:

a) Support for GPRS/UMTS QoS during SGSN change (inter SGSN RA update). Modification of the contemporary inter SGSN RA update mechanisms to become similar (if not converged with) GSM inter MSC handover type mechanisms.

b) Modification of the inter SGSN signalling mechanisms to support the transfer of related information directly between SGSNs (e.g. SRNC relocation parameters, cipher/security information).

c) Development of mechanisms to support single MM (the relation of updates between the MSC/SGSN and HLR is for further study). The Gs interface may be enhanced to support this capability.

Developments in contemporary GSM/GPRS network are also required to enable the UMTS <=>GSM/GPRS interworking since the anchor point is currently the GGSN.

5.9.3 The non-Anchor SGSN concept

The non-anchor SGSN concept may be viewed as the method currently used within GPRS (R97) for a change of SGSN.

5.9.3.1 Current GPRS operation

Current GPRS does not use an anchor SGSN (the SGSN used at PDP context activation may not be used by the MS during the lifetime of this PDP context).

The main reason is that, while in Circuit Switched GSM the call duration is very short, the PDP context duration may be very long (and the user be very far away from the SGSN where it activated the PDP context).

Furthermore, current inter-2G-SGSN mechanisms do not support a ‘drift’ SGSN since, at the reception of a downstream PDU, it is not possible to page a MS in standby state through another SGSN (there is no support of this requirement for a (R97) SGSN).

Note: When in (UMTS) RRC Connected mode, the UTRAN caters for paging of the mobile when in PS CONNECTED.

5.9.3.2 Developments of GSM/GPRS for the non-SGSN based anchor

To satisfy the identified requirements for GSM/GPRS/UMTS R99, the following developments will be needed to the contemporary GSM/GPRS network:

a) Support for GPRS/UMTS QoS mechanisms during inter SGSN RA update, this will involve continued linkage of the GGSN with the inter SGSN RA Update.

The current mechanisms for inter (2G)SGSN RA update are different to the mechanisms for inter MSC handover .

5.9.4 Analysis and comparison of the “anchor SGSN” and “non-anchor SGSN” concepts

The following aspects need to be considered when considering inter SGSN RA update concepts for GPRS/UMTS:

- Support of QoS requirements (e.g. transfer delay (for real time traffic), reliability (ability to handle correctly traffic requiring a high reliability), service interruption (for real time traffic)
- Relationship to mobility management
- Support of Class A/Simultaneous mode operation
- Resource usage within the network
- Developments needed within the standards

5.9.5 Support of QoS requirements

Transfer delay

Both the network and radio paths create delay within GPRS/UMTS communications. The non-anchor mechanism always crosses three GPRS nodes during communications (RNC, current SGSN, GGSN). The “anchor SGSN” architecture uses the same 3 nodes until an SGSN RA update occurs, then a new node (the drift SGSN) is added, with the communications ‘anchored’ at the initial SGSN. After an inter SGSN RA update in the SGSN anchor mechanism 4 nodes are used (RNC, drift SGSN, anchor SGSN, GGSN), the anchor SGSN relays user packets to the drift SGSN. The “non-anchor” architecture provides a lower network transfer delay and a lower jitter on this delay (less nodes implies less queuing). This is likely to be an issue for real time traffic such as VoIP. The impact of this needs to be assessed in relation to the delay over the radio path.

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Reliability

Within the UTRAN the (acknowledged) RLC layer between UE and SRNC provides the reliability required by some (non real time – high reliability) traffic within the UTRAN. When there is a change of SRNC:

- - either the RNCs (if there is no LLC in the protocol definition of Iu) using packet transfer between old and new RNC
- - or the CN (if there is an acknowledged LLC in the protocol definition of Iu) using packet transfer between old and new SGSN

can repeat the non acknowledged packets ensuring the reliability requested by the user.

The same reliability can be provided in both “anchor SGSN” / “non-anchor” SGSN architecture.

It should be noted that ARQ mechanisms (using acknowledged mode with repeats) do not guarantee to avoid break in transmission for real time applications (such as speech/VoIP).

5.9.5.1 Service interruption at SRNS relocation

With the anchor SGSN architecture service interruption may exist during the change over of path from old RNC to new RNC, mechanisms such as parallel paths could be used to prevent or minimise this. The anchor SGSN would acts as the anchor for multiple PDP contexts (potentially to different GGSN which could be located within different networks).

With the non-anchor mechanism service interruption may exist during the change over of path within the GGSN between the old SGSN and the new SGSN. The impacts (upon timing of inter SGSN RA update) of multiple PDP contexts (potentially to different GGSN which could be located within different networks) needs to be studied.

The impact on nodal buffering and path change requirements for both concepts

(e.g. between GGSN and old/new SGSN in non Anchor concept, and between anchor SGSN and drift SGSN in anchor concept), combined with the support of real time and non real time traffic needs to be assessed further.

5.9.5.2 Network resources used

As shown in Figure ~~12~~16, the non-anchor SGSN architecture requires less nodes and transmission resources than the anchor SGSN architecture. However, the impacts upon the network resources in terms of signalling, buffering and processing load requirements need to be addressed.

5.9.5.3 Quality of service requirements

The optimum mechanism to satisfy the service requirements need to be considered, for example for a non real time, long duration packet session the anchor SGSN may not be optimum. Alternatively, for a real-time short duration packet session the non-anchor concept may not satisfy the QoS requirements at SRNS relocation. However, if SRNS relocation is not performed for real-time short duration packet session, there is no break in transmission at all (It is acceptable since the duration is short).

A mix of solutions may need to be considered in relation to the Quality of Service requirements of the packet session.

5.9.5.4 Support for Class A (GSM/GPRS) and UMTS Simultaneous Mode operation

Within GSM/GPRS the mechanisms used within the MS and the network to support Class B/C operation are different to those required for Class A. Simultaneous mode is required within UMTS (R99) which will place requirements to the GSM/.GPRS/UMTS R99 standards. The impacts on the network and MS usage and control of radio resource need to be addressed.

5.9.5.5 Mobility Management

For MM point of view, interworking with 2G-SGSN has to be considered. A non-anchor SGSN architecture makes it easy since the GGSN is the anchor point in both 2G-GPRS and UMTS networks. The concepts chosen for UMTS and GSM/GPRS for R99 need to be compatible.

In the case SGSN anchor concept is introduced in R99 GPRS, several issues have to be considered:

- A new relaying protocol has to be introduced since BSSGP does not fulfil this requirement,
- The MS behaviour has to be modified: in standby state, it has to initiate a cell update instead of a RA update,
- The drift SGSN has to route the cell update to the anchor SGSN it does not already knows,
- When receiving a downstream PDU, the anchor SGSN has to page the MS under another SGSN. The use of P-TMSI may lead to conflicts since the same P-TMSI value may be already used for another MS.
- Interception and charging aspects, since the GGSN and the MS could be in different regions.

The current mechanism with UMM uses different mechanisms for PS and CS MM. The impacts of both mechanisms on GPRS MM/UMM and security/ciphering need to be addressed.

Within the anchor concept there are no RA updates as long as the MS has an active PDP context via anchor SGSN.

The non-anchor concept leads to RA updates with every change of SGSN; however, there is no RA update as long as the SRNS is not changed since the SRNS acts as an anchor point in the UTRAN.

The impacts of inter SGSN RA update for both anchor and non anchor solution in conjunction with location based services (such as SoLSA) needs to be addressed.

5.9.5.6 Comparison of developments needed within the standards for GSM/GPRS/UMTS R99

- R99 will include the support of QoS within GSM/GPRS and UMTS.
- Class A operation and UMTS simultaneous mode will be required for R99.
- The anchor SGSN concept would include the specification of drift SGSN and packet forwarding mechanisms.
- The non-anchor concept may need enhancement to satisfy the QoS concepts and will need development to ensure the interruption for inter SGSN RA update can be achieved within the QoS requirements.
- The changes and developments needed to GPRS R97 to satisfy these requirements as well as inter-working to pre R99 networks needs to be addressed.

5.10 Quality of service

- Application/End to end QoS
- QoS Segments (e.g. Radio, UTRAN, CN, Internet)
- QoS Mapping (between different segments/layers)
- Radio Access Bearers

- Resource management
- Interfaces/APIs between Application, TE, MT
- Charging of QoS aware applications

5.11 Others

5.11.1 GPRS/IP support for Multi-media service

The following developments are needed within IP/GPRS to support the expected multi-media requirements of UMTS (note this list is not exhaustive):

QoS for GPRS: To enable real-time 'streaming' developments.

Adoption of IP Telephony, H.323 and equivalent PSTN/Internet technologies: To support the control and interworking of multi-media and telephony applications with non-UMTS networks.

Interrogation of the HLR with the Gateway functionality: To enable terminated communications to be delivered to the mobile terminal. This can include the PIG and H323 functionality.

Figure ~~7~~ 17 illustrates a potential architecture which could be used to deliver telephony and multi-media features.

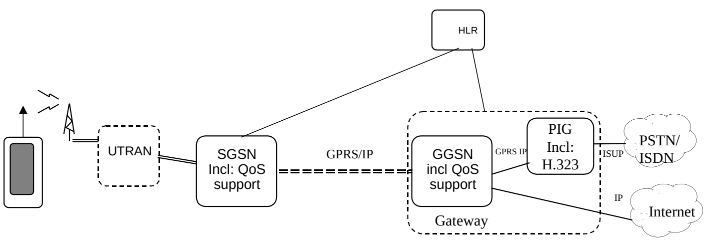


Figure 17~~3~~: Evolved GPRS/IP support for Multi-media services

Telephony and multi-media requirements for UMTS may be supported via the evolved IP/GPRS network of Figure ~~7~~17. This architecture does not need a separate non-IP based circuit switched (MSC) platform.

- Multimedia service control
- Phasing. What is for release -99, -00, etc. ?
- Network migration
- Handling and type of coded speech over Iu
- Location of ciphering functionality
- Link access control for user data (LAC-U)
- Data compression
- Allocation of resources of Iu

5.11.2 Separation of switching and control

Proposed Architecture

In this section the concept of a logically Separated Call Control (SCC) server is introduced. Currently CC is integrated with each of the MSCs in a network. Here it is suggested (and shown in figure 18) that a single CC function is implemented which is logically separated from the switch. The physical location of the SCC server is an implementation issue. Examples of implementation include:

- SCC integrated with IWU.
- SCC integrated with one or more switches. In this case a IWU may not be required between the SCC and the switch(s) with which it is implemented. IWUs would be used to connect to other switches.
- Standalone SCC.

Data required by the SCC could be held locally so as to reduce signalling load. This is likely to include data currently held in the HLR and VLR and a network resource database which allows the SCC to determine what network resources are available and record the state of resources e.g. used, reserved or free. Figures ~~2~~ 19 and ~~3~~ 20 show the signalling flows in the network for mobile originated and mobile terminated calls respectively.

In figure 1 MM is shown as being integrated with the SCC. It could equally well be separated from the SCC.

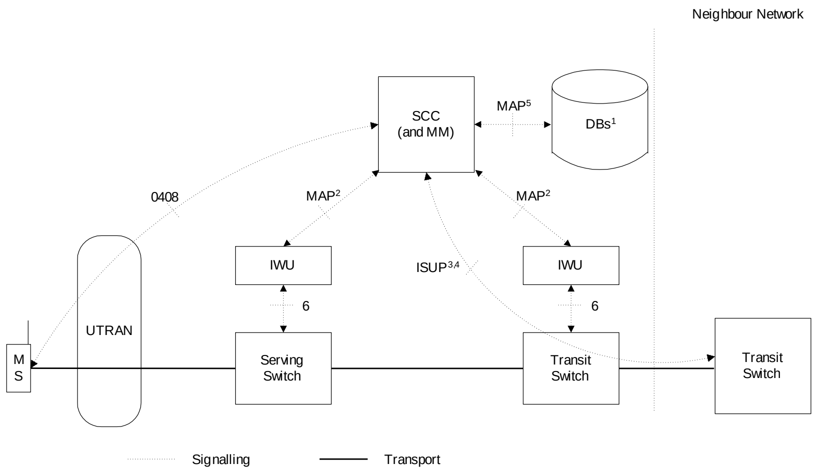


Figure 184 Network Architecture

Notes:

1. DBs represents all databases necessary for SCC operation, e.g. HLR, VLR, Network Resource database.
2. MAP (with some new operations) could be used here which would probably represent minimum change. Alternatively a more general protocol such as MGCP could be used which would represent more change but have the advantage that the switch would be made more generic.
3. ISUP (with some modified messages) could be used to communicate between the SCC and the transit switch of a neighbouring network.
4. This signalling is shown to pass through the transit switch as this is a likely (but not mandatory) route for it to take. The logical connection is between the SCC and the neighbour network.
5. MAP (with some new operations) could be used here which would probably represent minimum change. Alternatively a more general protocol could be used which would represent more change but have the advantage that the databases would become more generic. If an IN implementation is adopted the SSF could form part of the SCC which could communicate with an SCF via MAP or INAP which in turn could communicate with the DB via DAP.

6. This interface could be the same vendor specific propriety interface that is implemented today internally to the MSC.

Mobile Originated Call

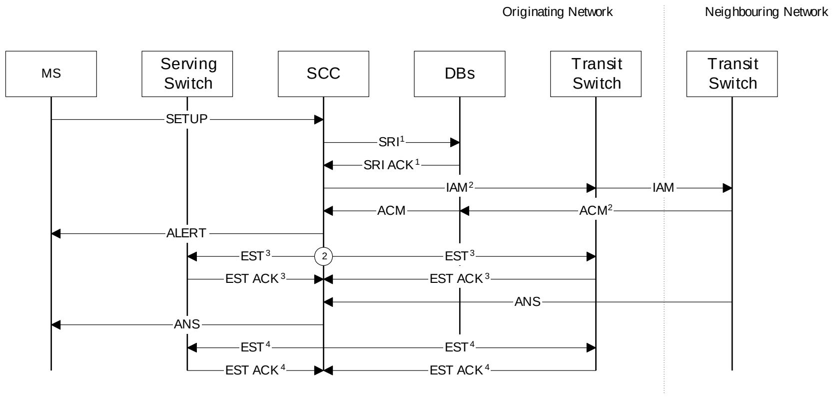


Figure ~~15~~19. Signalling Flow for MO call.

Notes:

1. A modified SRI operation could be used by the SCC to request routing information from the databases. The response contains all the information required to route the call from the serving switch to the point of interconnect.
2. A modified IAM and ACM could be used to communicate between the SCC and the transit switch. Because the SCC serves multiple switches a switch ID (in addition to a route ID and circuit ID) is required.
3. EST (establish) and EST ACK could be a new MAP) or could be provided by a new protocol such as MGCP. Here EST is used to instruct the switch to establish the backward connection. EST ACK confirms that the required connection has been established. Note that the SCC executes the EST operation to all involved switches simultaneously. In the event of a handover the SCC would execute EST operations only to those switches involved in the handover. In the event that the neighbour network is not controlled by an SCC the transit switch is unlikely to be involved.
4. Here EST is used to instruct the switch to establish the forward connection.

Mobile Terminated Call

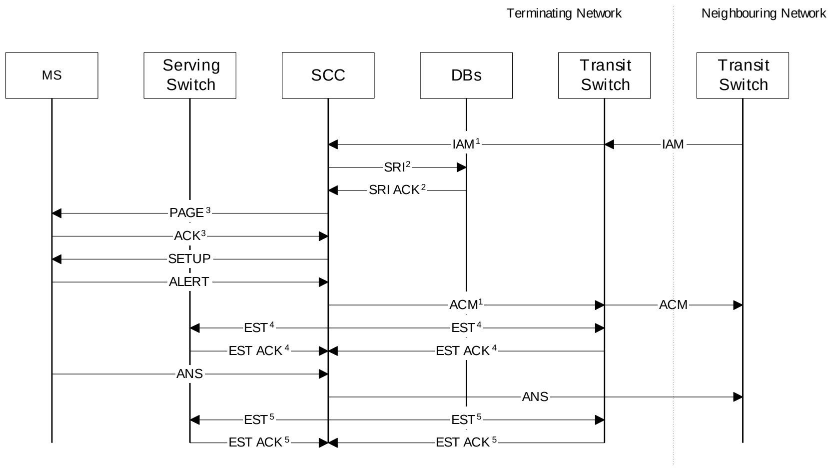


Figure ~~16~~20. Signalling Flow for MT Call.

Notes:

1. A modified IAM and ACM could be used to communicate between the SCC and the transit switch. Because the SCC serves multiple switches a switch ID (in addition to a route ID and circuit ID) is required.

2. A modified SRI operation is used by the SCC to request routing information from the databases. The response contains all the information required to route the call from the serving switch to the point of interconnect.

3. For clarity MM is considered as part of SCC here.

4. EST (establish) and EST ACK could be a new MAP) or could be provided by a new protocol such as MGCP. Here EST is used to instruct the switch to establish the backward connection. EST ACK confirms that the required connection has been established.

5. Here EST is used to instruct the switch to establish the forward connection.

5.11.2.1 Benefits

The separation of switching and control functions offers the following benefits:

Architectural Flexibility: The separation of bearer from the control allows flexibility in locating the desired functions.(functions could either be centralised or distributed). For instance, the switching and call control functions performed by a circuit or packet switch can now be separated and located in physically distinct locations. The control functions (all or a part thereof) could be located in a “call control server”, which can provide the information necessary to appropriately route the bearer. Further, this allows the use of platforms designed specifically for the task being performed to be used. Dedicated platforms will allow easier and faster software development and less work will be involved in rolling out new software versions.

Efficient Utilisation of Network Resources Given that most of the traffic associated with a call is bearer traffic, optimal routing of bearers (facilitated by separating control from the bearer) allows efficient utilisation of network resources. For instance, call control may be routed to a “ Call Control server” for purposes of address resolution, billing, enabling of services, and others, but the bearer does not have to traverse through the call control server. Further, optimal routing can be maintained during mobility (the concept of an anchor MSC can be removed)since the bearers can be re-routed after a change in location. Optimising the routing in this way will have greater significance for UMTS calls which are likely to be high bandwidth and may also consist of multiple streams.

- Further optimal routing can be achieved in the case of call divert.

Bearer Flexibility and Robustness: The separation of bearer and control allows the communicating parties to negotiate the resources required (even possibly re-route the bearers) even after call setup has been completed. Bearers could be re-routed during a call due to a change in the performance required or to work around failure of network elements.

Future-Proof: The separation of bearers from control makes the protocols used more modular (than before). For instance, the same control protocols can be used over multiple transport technologies. Further, the same control protocol can be used for establishing multiple bearer types. This facilitates improvements in technologies being used with minimal impact.

5.11.2.2 Drawbacks

Separation of switching and control means defining the interfaces between the various control functions (such as cal control, mobility management, session control, etc.) and the switching functions (i.e., switching matrix). For example in the case of a GSM MSC, this would mean defining an open interface between the MSC service switching functionality and the TDM switching matrix.

For packet data nodes, the separation might be more realistic as a client-server type of architecture is more natural in that domain. However, this is a deviation from the current GPRS and therefore may require additional standardisation effort.

5.12 New Handover functionalities

The radio access network has to be capable of connecting to a variety of existing core networks. This leads to a requirement that the UTRAN will be allowed to connect with evolved forms of existing CNs. There will be the need to support new Handover functionalities between UMTS and 2G systems.

The support of multimedia services and the separation of Call Control and Connection Control (many connections: telephony, video, data could be associated with one single call and handed over separately), together with a micro or pico-cellular environment will cause increased complexity of Handovers compared with GSM.

Developments will be needed of the contemporary GSM/GPRS platforms to enable handover/cell reselection of communications between GSM/GPRS and UMTS. To enable this specific developments are needed for:

- Handover/cell reselection of communications which have inherent delay and error requirements (e.g. speech as for contemporary GSM circuit switched and speech/ video).
  (This may be viewed as an equivalent of GSM circuit switched handover).

- Handover/cell reselection of communications which may not have inherent delay requirements but do have error requirements (e.g. packet data communications such as IP/GPRS, file transfer, SMS).
  (This may be viewed as an equivalent of GPRS cell-reselection).

This also requires the ability to potentially ‘negotiate’ and modify communications parameters when handing over between GSM/GPRS and UMTS.

- It would be useful to provide new procedures in UMTS in order to make handover a totally Radio Resource Management procedure fulfilled as far as possible by the BSS without the intervention of the NSS part. The proposed interconnection of BSSs to allow for handover streamlining could be a step in this direction. (This may be difficult when performing hand-over between different environments, and a traditional GSM-like handover procedure is likely to be used in this case).

- It is likely that the network performance during handovers will be increased by restricting handover to the access network, leaving the core Network to deal with the Streamlining procedure without any real-time constraints. (In the case of a successful GSM inter-BSC handover, eight messages are exchanged real time on the A interface between the MSC and the two BSC; if Streamlining is used, this could be potentially reduced to two messages (Streamlining Request - Streamlining Acknowledge) with a significant saving in the signalling overhead.

As part of the overall QOS negotiation between user and network, mechanisms will be needed to enable parameters such as handover delay, jitter, packet/information loss/acceptable error etc. to be applied as part of the communications path requirements utilised during the communications ’session’.

A number of options are available to support handover within the UMTS Core Network; real time support within the core network, real time handover within the UTRAN with subsequent ‘streamlining’. Irrespective of the final mechanism developed within the UMTS Core Network for UMTS handover, functional developments are needed within the Core Networks (both GSM/GPRS and UMTS) to support handover between UMTS Core Networks and evolved GSM/GPRS core networks.

~~5.13 Reduction of UMTS signalling~~

~~5.13.1 Turbo Charger~~

~~The signalling load associated with subscriber roaming can be high when either the location areas are small or the subscriber travels significantly. The Turbo-Charger concept aims to optimise signalling associated with subscriber data management by assigning one MSC/VLR to perform the Call Control and Mobility Management functions while the subscriber remain attached or until signalling routes require further optimisation.~~

~~The benefits of the Turbo-Charger concept are:~~

- ~~the substantial reduction in signalling traffic for subscribers located in the home PLMN,~~
- ~~the substantial reduction in signalling traffic between the visited PLMN and the home PLMN,~~
- ~~no new network nodes are required,~~
- ~~applicable to a wide range of protocol used for the transfer of data.~~

~~The disadvantages of the turbo-charger concept are:~~

- ~~Connections are required from the access network to be fully meshed to all MSCs in the turbo-charger area.~~

~~5.13.1.1 Overview of the Turbo-Charger Concept~~

~~A Turbo-Charged network constitutes a network architecture designed to reduce mobility management costs and provide automatic load-sharing between MSC/VLRs.~~

~~The architectural philosophy is to equally divide the subscribers between the available MSC/VLRs, irrespective of their location. In the context of GSM, this could be achieved by placing a routing function (e.g. evolved STP) between the BSC and the pool of MSC/VLRs. The purpose of the routing function is to route A-interface messages to the MSC/VLR that is serving the mobile station. The solution requires the MS to store a discriminate that can be used to identify the serving MSC/VLR and for routing to be applied on this discriminate on the connection between the MSC/VLR and access network. A TMSI partitioning scheme could be utilised. This scheme allocates a sub-set of the TMSI range to each MSC/VLR, Figure X. The A-interface messages are then routed to the right MSC based on the TMSI. This could be done by a routing function external to the access network implying no access network modification (see figure x). If a TMSI partitioning scheme is used then new SIM cards are not required.~~

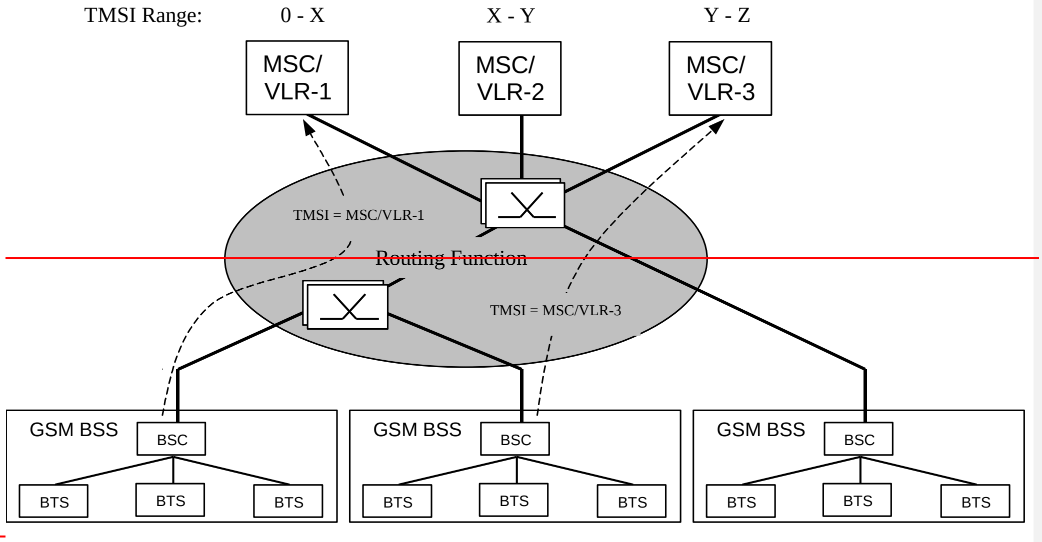
~~The temporary identity used for paging (TMSI) must be unique within all the MSCs in the turbocharger area. This implies that there must be a mechanism to ensure that this requirement is met for turbocharged MSCs (e.g. TMSI partitioning).~~

~~Two mechanism to provide load-sharing are envisaged, random load-sharing and dynamic load-sharing.~~

~~Random load-sharing requires the routing function to randomly assign a MSC/VLR to serve a particular mobile station when it first comes in to the network. Regardless of where the mobile is the same MSC/VLR will always serve it provided the mobile remains in the area served by all the turbocharged MSC/VLRs linked by the routing function.~~

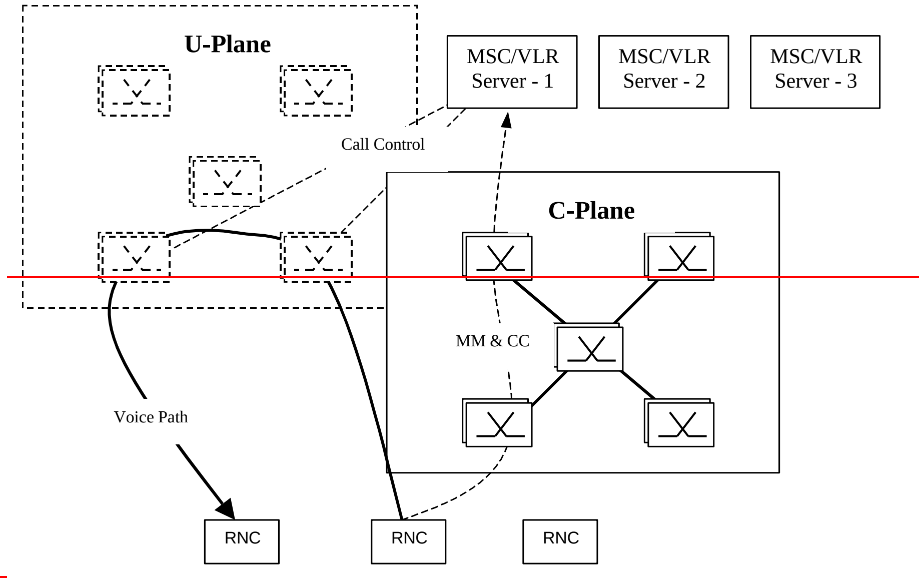
~~In large metropolitan areas where subscribers are served by multiple MSC/VLRs, some MSC/VLRs may be very busy while others are not fully utilised. Dynamic load-sharing requires the implementation of an intelligent router. Since the routing function routes all A-interface traffic, it can participate in load-sharing and balancing based on the current loading of each MSC however linkage between MSC load and the routing algorithm would be required.~~

~~In the case of a Turbo-Charged network where the network is sub-divided into large regions, further optimisation can be achieved by adding the Super-Charger functionality.~~



~~**Figure 17: Example of GSM Turbo-Charger Network Architecture**~~

~~In the context of UMTS, the routing function becomes a feature of the RNC, see **Error! Reference source not found.**~~



~~**Figure 18: Example of UMTS Turbo-Charger Network Architecture**~~

~~5.13.2 Relationship between GLR and TurboCharger~~

~~The GLR and TurboCharger are two independent schemes for reducing the amount of MAP traffic generated in UMTS networks.~~

- ~~The GLR works by reducing traffic between PLMNs associated with Location Updates. This is achieved by "caching" the roaming subscriber's data in the visited network~~
- ~~The TurboCharger works by eliminating the need to perform location updates. The same VLR can hold a subscriber's data for the duration of his attachment to the network.~~

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~~A TurboCharged network requires that each MSC/VLR can physically connect to all RNCs. Therefore TurboCharging may be best suited to areas of the network characterised by dense geographic coverage. On the other hand, the GLR function is independent of the network density.~~

~~The network structure illustrated in **Error! Reference source not found.** shows that the GLR and a TurboCharged area within the same PLMN are independent. In fact, it shows benefits from using the two techniques in the same network. The Turbo-Charger reduces the location registration signals between the MSC/VLR and GLR:~~

- ~~There is no new update location signal between MSC/VLR and GLR if roamer moves inside of the Region A.~~
- ~~There is no new update location signal between GLR and HLR if roamer moves between regions.~~

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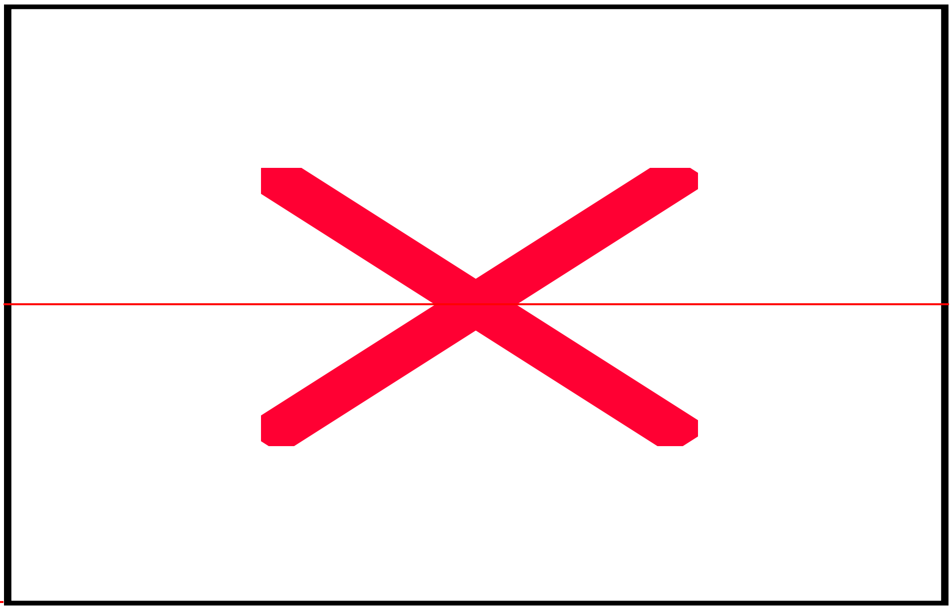


~~Figure 19.~~

5.14 Transcoder Control

In order to improve voice quality for mobile-to-mobile calls (MS-MS calls) in GSM Phase 2+ networks, Tandem Free Operation (TFO) using in-band signalling has been specified. The equivalent function in Japan's PDC (Personal Digital Cellular) network is known as Transcoder Bypass, which has been specified to make use of out-of-band signalling control (i.e. by the PDC-MAP protocol).

It is likely that UMTS terminals will support a wider range of codecs than is currently the case for GSM terminals. In the case of calls between UMTS terminals, codec negotiation will be needed to:

- match terminal capabilities during call establishment

- support supplementary services interactions such as with conference/Multi-party calling, ECT, CFNRy
- support changes in radio interface conditions.

This requires control of the transcoder unit in the UMTS Core Network during (and after) call establishment and handover. However, the inband signalling technique currently specified for GSM-TFO has limitations in this area. For example:

- **UMTS call setup;** the GSM-TFO mechanism is designed to support a limited set of codecs. Each time a new codec is introduced into UMTS the transcoder would need to be upgraded.
- **UMTS call in progress;** codec negotiation using the GSM-TFO mechanism would need complex in band signalling.

The different solutions to support the required functionality for transcoder control in UMTS need to be studied in detail. Signalling for codec negotiation and control may be achieved by:

- New control mechanisms between the mobile terminal and the network based transcoder (out of band and in-band solutions need to be studied).
- Revisions to ISUP signalling.
- Revisions to MAP signalling.
- Inband signalling mechanism developed for AMR

It is for further study what impact transcoder control has upon networks external to the PLMN.

5.15 Support of multimedia services

One of the most important requirements for UMTS is the capability of supporting multimedia services.

The following principles should guide and apply to the support of multimedia services in UMTS:

- Multimedia services in relation to UMTS should be standardised and handled according to emerging multimedia standards. SMG should not standardise multimedia services solely for UMTS networks. SMG should take advantage of existing and emerging main stream standards for multimedia, in reality defined outside of the UMTS.
- Multimedia applications according to such main stream standards should be *supported* (transported and handled) *efficiently* in the UMTS.
- Multimedia requirements on the UMTS should, as far as possible, explicitly be related to such multimedia application standards to be supported – rather than to generic statements or assumptions related to the architecture.
- The multimedia bearer capability requirements, incl. QoS, are expected to effect the core as well as the radio network.

Among others, two requirements for an efficient support for multimedia applications, which currently can not be achieved by GSM, are sufficient bandwidth allocation and flexibility of bearers.

- The bandwidth requirement relates to the transport technology used on (both the radio and network sides). In particular switching and transport capabilities within the network must be able to support, in an efficient and flexible way, air interface rates of at least up to 2 Mbit/s. It is unlikely that a 64 kbit/s based switching system will be able to do this in the most efficient manner.
- Separation of call control from connection and bearer control. This is an important requirement to satisfy the concept of Quality of Service for media components: a call/session may use various connections at any one particular instant (making use of one or several bearers). It should then be possible to add or remove bearers

during such a call in order to cope with user needs or problems on the radio path. (Ref. ETS 22.01 Service Principles)

5.16 Support of services requiring variable bit rate

- If a number of applications use VBR data flows then packet transfer mode on the radio and network side has to be considered in order to make efficient use of resources.
- If packet transfer is allowed on the radio side, a finer degree of location management is/may be needed for radio resource optimisation (if only the LAI is used as in GSM, packets addressed to one single mobile terminal would need to be broadcasted over its entire Location Area; a new routing concept playing a similar role to the GPRS routing area is then needed).These additional Radio Resource/Mobility Management functions could be located in the Radio Access Domain, containing data strictly related to the access techniques that could be hidden from the serving network.

5.17 UMTS Simultaneous Mode.

Within GSM/GPRS Class A mobiles have been defined which support 'simultaneous operation of both GPRS and other GSM services'. UMTS is intended to enable users to access a variety of communications features including access to PSTN/ISDN services/features as well as IP capability. The UTRAN developments have a mechanism to support common 'pipe' over the radio interface (the RRC Connection). It is expected that multi-media and mixed media (PSTN/ISDN/IP communications) will play a large part of UMTS communications.

From this perspective it is essential for UMTS that from day 1 of network launch that mobile terminals can support both PSTN/ISDN services and features as well as IP simultaneously. Based upon this aspect 'Simultaneous mode' has been defined for UMTS communications. This definition can be applied to both network and mobile terminals.

Simultaneous mode is defined as the support of active parallel CS and PS communications.

The UE has simultaneous PS MM Connected and CS MM Connected states when in UE simultaneous mode.

Note: The support of 'Simultaneous mode' should not prevent the operation of mobile terminals in solely CS MM or PS MM connected mode. Simultaneous mode capable terminals should be supported in CS service and PS service only capable networks. Operators may wish to just use 3G_MSC and/or 3G_GSNs if required.

The impact of supporting 'Simultaneous mode' operation of the UE needs to be addressed within the UMTS System as a whole. In particular, the impacts upon the UE, radio, UTRAN and Core Network nodes need to be assessed.

5.18 GSM and UMTS cells in the same registration area

The concept of GSM and UMTS cells in the same registration area was introduced in order to minimize location update signaling when changing between GSM and UMTS systems. Especially, the lack of UMTS coverage, e.g. in-building coverage in urban or suburban areas, can lead mobiles frequently changing between GSM and UMTS systems. In such case, the common registration area concept is anticipated to reduce the signaling compared to GSM and UMTS cells in different registration areas.

Currently GPRS routing area updates cause serious disruption to the user plane traffic. This was acceptable in GPRS only because routing area updates would be rare events which should only occur at the borders of large, well planned geographic areas.

If GSM and UMTS routing areas are overlaid, it can be expected that many mobiles will change routing areas at rates greater than in GPRS. In such cases the (negative) impact on the quality and throughput of the data will be very significant. Placing UMTS and GSM cells in the same RA greatly alleviates this effect.

If the core network does not use the Gs interface then, while the mobile is performing a routing update in the GSM cell the mobile will not be pageable from the MSC. Frequent routing area updates will have a serious effect on mobile terminating call success rates. Note that implementing the Gs interface only really helps in the case that the GSM and UMTS cells use the same SGSN. Therefore, this pageability problem disappears when the UMTS and GSM cells are in the same RA.

Third generation needs to offer higher quality (eg higher MT call success rate) than second generation. Hence the capability to have GSM and UMTS cells in the same Registration Area is needed for at least CS traffic.

5.18.1 Open issues

The following open issues, opportunities and challenges have been identified concerning this concept:

a) Security.

In UMTS both MS authentication and network authentication is planned to be implemented. Some solutions have been proposed for MS authentication, ciphering and integrity check during change from GSM to UMTS but not any for network authentication. However, all these issues need to be resolved in order to perform handover between UMTS and GSM (either on the MSC, or, on the SGSN side, or both) so it is a requirement that the issue is solved.

b) Network service capabilities.

The availability of network service capabilities should be indicated to the end user somehow in order to offer confidence in the cellular service. However, any communication that starts in GSM cells needs to be easily transferable to UMTS cells, and vice versa. Usage of a UMTS cell does not guarantee any particular data rate: the data rate will vary with, at least, range, load on the cell and interference levels. Thus merely being camped on a GSM or a UMTS cell will give no real indication of the services available from that PLMN - this is irrespective of whether or not the cells are in the same or different registration areas.

The most likely source of 'network service capability' information will come from the core network. As the mobile contacts the core network whenever it does a registration update, it seems sensible that we allow network operators the option to send "network service capability" information in MM (and GMM) messages sent to the mobile.

c) Terminal capabilities

Terminal capabilities have some affect on the service availability e.g. in a situation where a dualmode terminal makes an attach in UMTS and later on moves into GSM cell. If mobile is PS and CS attached in UMTS, it may need to detach either CS or PS in GSM (e.g. class C mobile).

d) Idle mode control

Within a particular visited network it can be expected that the Core Network will restrict some subscribers to only UMTS cells, restrict others to only GSM cells, and permit some to use both types of cell.

Mobility Management signalling (MM and GMM) needs to be developed so that the (dual mode) mobiles in idle mode adapt their cell reselection procedures according to the Core Network's instructions.

e) Paging channels

When GSM and UMTS cells are in the same common registration area, the overall paging channel capacity is given by the minimum of the GSM paging channel capacity and the UMTS paging channel capacity. If the capacity of the two channels cannot be configured so that the UMTS paging capacity is larger or equal to that of the GSM channel, then paging capacity will be wasted. Note that UMTS paging channels with larger capacity than GSM paging channels probably do not cause any inefficiency (because under occupancy of the UMTS paging channel probably only leads to less radio energy being transmitted). Hence it is expected that UMTS paging channel capacity need to exceed or equal that offered by the GSM radio interface (in the cases of a GSM combined control channel and the case of a single GSM non-combined control channel). Note that this is relevant for both CS and PS domains.

In any case, in a common registration area GPRS pages are always sent also via UMTS paging channel, due to the nature of GPRS paging (CN paging). This may have some impact on paging channel capacity needed in UMTS.

5.19 Short Message Service Cell Broadcast in UMTS

The *Short Message Service Cell Broadcast* (SMS.CB) was defined as a UMTS Phase 1 requirement to guarantee the continuity of the corresponding GSM services. It shall be provided seamlessly (as far as the user or the users terminal equipment is concerned) across the UMTS and GSM network.

5.19.1 Network Architecture

Figure 22 proposes a straight forward adoption of the GSM cell broadcast architecture in UMTS.

The basic network structure replaces the GSM BSS with the UTRAN containing the RNC and the Node B. The cell broadcast center (CBC) is part of the core network and connected to a routing node e.g. a 3G SGSN via the Bc reference point. Thus the CBC can reach every RNC via the user plane of the Iu interface by using the newly introduced common communication channel. On the logical interface between the CBC and the RNC a mandatory protocol shall be defined. which should mainly be adopted from the corresponding GSM specification (see GSM 03.41). The other UTRAN related interfaces are described in the according UTRAN specifications based on the RAN 2 TR 25.925. Based on this architecture and the current requirements for cell broadcast the core network elements like MSC, VLR, HLR etc are not involved for the service delivery.

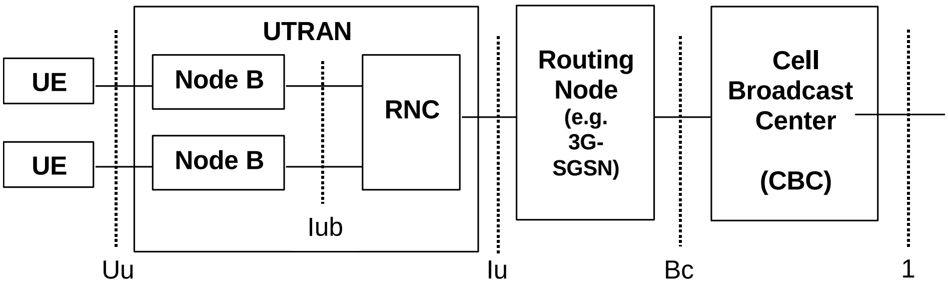


Figure 21: Architecture for SMS Cell Broadcast in UMTS

The protocol stack between the CBC and the RNC is given in figure 22. Protocol primitives for the cell broadcast application defined by GSM 03.41 are used for the Cell Broadcast application.

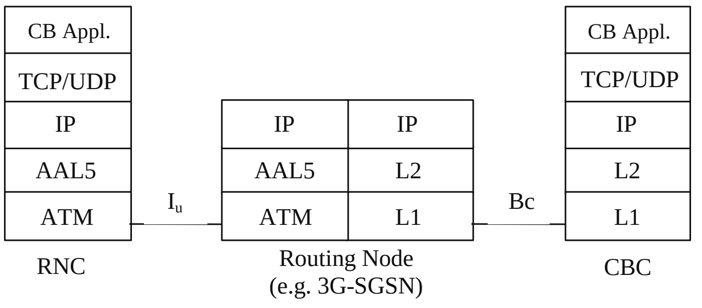


Figure 22: Common Communication Channel used by the Cell Broadcast Application

5.19.2 Interface Responsibilities

Interface 1 was not in the scope of GSM (see also GSM 03.41). At the moment it is ffs. if it should be standardized.

The interface between the CBC and the RNC is in the scope of T2 SWG3 (Messaging) as this group is continuing the work of the SMG4 Drafting Group *Message Handling.* Work has not yet started.

The needed changes to the Iu and Iub Interfaces is in the scope of RAN WG3 mainly.

The Uu Interface is fully under scope of RAN WG2 for layer 2 and 3 and RAN WG1 for layer 1 questions.

5.19 Use of Single MSISDN for CS and PS Voice Services

5.19.1 Introduction

Within release 99 it is a working assumption that existing and future multimedia protocols can be supported by the UMTS CC/SM as application layer protocols. Where terminals support voice over both CS and PS domains, MT calls currently would require separate MSISDNs. It is desirable to allow the use of a single MSISDN and allow the network to determine how to route the calls. The mechanisms described below allows the use of a single MSISDN. The mechanism shall not restrict the network to only supporting a single MSISDN.

5.19.2 Option 1

In overview, when an incoming call arrives from the PSTN, GMSC (gateway MSC) sends an enquiry, via HLR, to the VLR for the MSRN in order to route the call request to the serving MSC. Upon receiving this enquiry from HLR (containing the MSISDN of the called UE), VLR should be able to assign a MSRN based on the decision on how to terminate the call, i.e., either via CS or PS domain. The MSRN assigned will be different (in prefix, for example) so that the ISUP message can be sent to the serving MSC or the serving GK correspondingly.

In order to provide the capability for a user to be contactable for voice services via CS or PS domain using the same MSISDN, two enhancements are for-seen to the VLR.

- VLR needs to be able to store the UE's capability as a VoIP terminal, and user's preference of accepting a call via VoIP or CS call.

- VLR should be able to assign a MSRN based on the decision on how to terminate the call, i.e., either via CS or PS domain. The MSRN assigned will be different (in prefix, for example) so that the ISUP message can be sent to the serving MSC or the serving GK correspondingly.

Furthermore, each GK shall be associated with a single VLR, and shall be required to support a MAP-B (VLR-MSC) like interface. A VLR may however be associated with multiple GK.

Figure X illustrates in greater detail how the single number and path selection can be supported. It shows a scenario of incoming call from PSTN/ISDN domain to a roaming UE. The message flow is as follows:

1. Call request from ISDN in ISUP message, which contains MSISDN of the called party.
2. GMSC gets the request and issues a enquiry to HLR.
3. HLR, knowing the called UE is roaming, issue a enquiry to the VLR in the visiting network.

4. The VLR replies with a MSRN, which is associated with GK/Signalling-Gateway interface or MSC interface, depending on the path selected.

5. HLR relays the MSRN back to GMSC.

6. GMSC continues to route the call to the MSC or GK/SG in the visiting network:

   6a: if the CS path is selected (reflected by the MSRN returned), the call is routed to the MSC. The message flows after that is not shown.

   6b: if the PS path is selected, the call is routed to the GK/SG via PSTN/ISDN. The GK will then setup a call to the UE over PS domain via a PSTN/IP gateway. The message flow between GK and UE is not shown.

7. If the PS path is selected, the GK shall contact the VLR in order to provide the mobile identity (IMSI) for the MSRN.

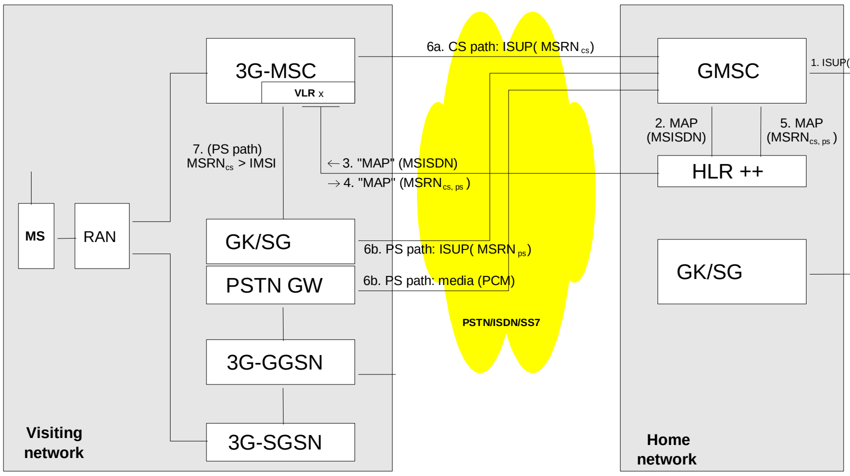


Figure 23

Step 7 could further be enhanced to provide subscriber information pertaining to supplementary services, hence allowing common services to be provided to the user, regardless of the path (CS or PS) used to route the call.

5.19.3 Option 2

When an incoming call arrives from the PSTN, GMSC (gateway MSC) sends an enquiry to get a MSRN in order to route the call request to the servingnode. Upon receiving this enquiry from GMSC (containing the MSISDN of the called UE), HLR should be able to determine VLR that will assign a MSRN. This HLR determination of the VLR is based on the decision on how to terminate the call, i.e., either via CS or PS domain. The MSRN assigned will be different (in prefix, for example) so that the ISUP message can be sent to the serving MSC or the serving GK correspondingly.

In order to provide the capability for a user to be contactable for voice services via CS or PS domain using the same MSISDN, two enhancements are for-seen to the HLR.

- HLR needs to be able to store
  - the user's rights to use speech services using CS and / or PS domain

- user's preference of accepting a call via VoIP or CS call.

- Upon user's subscription and preference and upon the registration status of the UE (either registered on a MSC, on a GK or on both), the HLR chooses a VLR (either VLR of the MSC or VLR of the GK) whom to request a roaming number from. If due to user detach or due to internal load, the first chosen VLR does not allocate a roaming number, then the HLR requests (if allowed by user's subscription) a roaming number from the VLR of the other domain (if the address or the VLR of the other domain is known to the HLR)

Figure X illustrates in greater detail how the single number and path selection can be supported. It shows a scenario of incoming call from PSTN/ISDN domain to a roaming UE. The message flow is as follows:

1. Call request from ISDN in ISUP message, which contains MSISDN of the called party.
2. GMSC gets the request and issues an enquiry to HLR.
3. HLR, knowing where the called UE is roaming as well as the user's subscription and preferences, determines the VLR associated with GK/Signalling-Gateway interface (3b) or MSC (3a) in the visiting network and issues an enquiry for a roaming number.
4. The VLR replies with a MSRN. (4a if the VLR is the VLR of the MSC, 4b if it is the VLR of the GK)
5. HLR relays the MSRN back to GMSC.
6. GMSC continues to route the call to the MSC or GK/SG in the visiting network:

   6a: if the CS path is selected (reflected by the MSRN returned), the call is routed to the MSC. The message flows after that is not shown.

   6b: if the PS path is selected, the call is routed to the GK/SG via PSTN/ISDN. The GK will then setup a call to the UE over PS domain via a PSTN/IP gateway. The message flow between GK and UE is not shown.

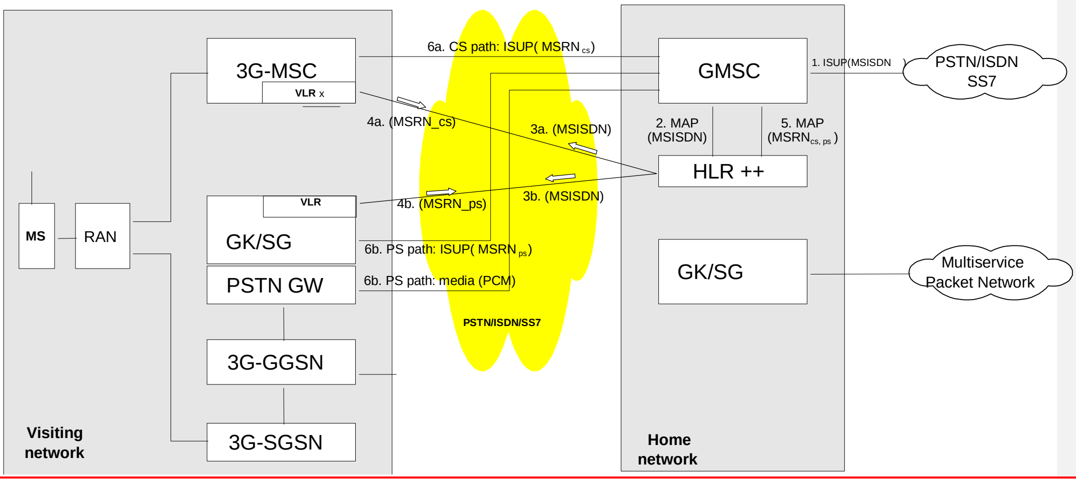


Figure 24

The VLR of the GK had previously updated its location to get subscriber information pertaining to supplementary services, hence allowing common services to be provided to the user, regardless of the path (CS or PS) used to route the call and to give its address to the HLR.

5.19.4 Option 3

This chapter describes an architecture and exemplifying call flows for inter-service roaming for Telephony as classical TeleService Speech in GSM/UMTS networks and Telephony as the voice component of a MultiMedia service. The chapter is intended to show the reasons and principles of the proposed architecture.

Terminology

- **Personal Number Domain.** An optional architecture that is overlaid over the GSMisdn Domain and the IP MultiMedia Domain to support user reachability according various criteria such as Network domain attachment(s), user preferences, incoming traffic characteristics etc.
- **Personal Number Service.** The service supporting reachability across Network Domains.
- **Personal Number Function (PNF).** The function performing the routing control.
- **InterroGation Function (IGF).** A call processing function that routes call to the appropriate Network Domain based on interrogation of the Personal Number Function.

Architectural overview:

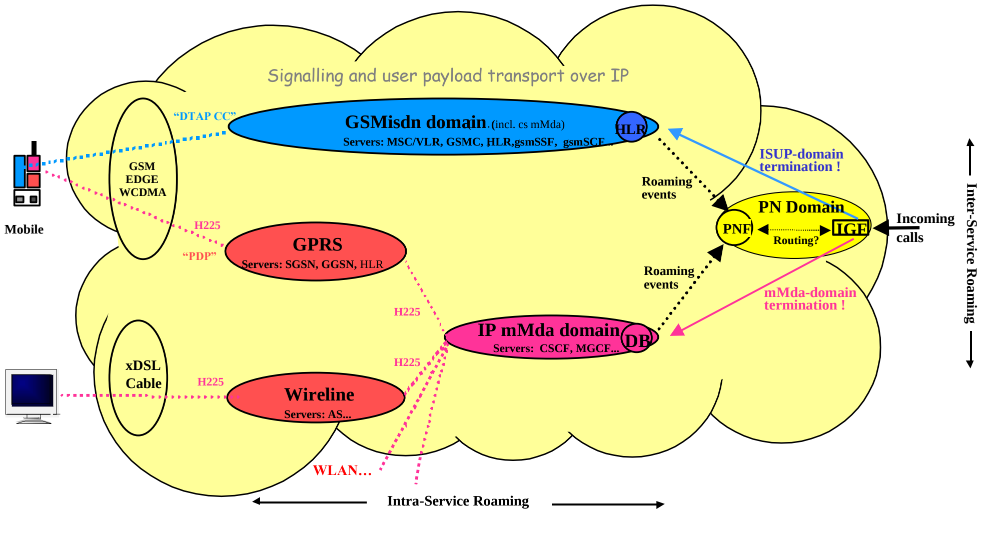


Figure 25: The Personal Number Domain architecture

Here it is proposed to provide this functionality overlaid as a Personal Number service, typically occurring in the home environment. The reason for this is to keep as loose as possible coupling between the GSMisdn Domain and the IP MultiMedia domain. The aim is to avoid disturbing the user experience in the tremendous growth in GSMisdn Domain deployment and to avoid restrictions and delays for the development of the IP MultiMedia domain. The need for the functionality may vary as coverage for the IP MultiMedia service becomes more complete- looking to a distant future, this overlaid routing function may not be needed at all. Seen from the all IP network, the GSM Speech fall-back could be seen as an architectural exception, however for years to come commercially very important.

Based on these extensive flexibility requirements we propose the overlaid approach rather than to implement the functionality into the core of both Network Domains. Thus we have rejected any solution where the IP MultiMedia Domain regards the GSMisdn network as a visited network due to associated complex mappings user profiles. The reason is the necessary freedom for new innovative add-on services within the IP MultiMedia Domain which would become impossible to map transparent enough into GSMisdn services.

In line with the VHE/OSA approach, we propose to implement capabilities in both domains to report routing impacting events to the Personal Number Domain. This may also be complemented by direct (network transparent) interaction from the mobile or the user.

This approach allows for maximum control power to the home environment on the design of the routing algorithms, to make them customised per user, to change them by time etc. Even if use of a single reachable number creates the need, nothing restricts using the concept for other cases as well and even expand it to a full fledged add-on services/applications concept, as desired by each operator

It is also assumed that the Personal Number Domain is regarded optional in the architecture. e.g. for a situation where there is not need for a fall-back from the IP MultiMedia service to the GSMisdn Domain Teleservice Speech, the discrimination of users to various IP accesses shall also be possible within the IP Multimedia domain.

The Personal Number Domain is suggested as a call processing Personal Interrogation Function linked into the call chain (thus not excluding sequential or parallel hunting algorithms) and a Personal Number Function hosting the user preferences, visited network domain(s) etc. as well as the algorithms for routing decisions. Thus, the service can be introduced with minimal coupling to the Network Domains. The needed adaptation of the served underlying networks is the capabilities to report routing impacting events to the Personal Number Function. This approach is aligned with the VHE/OSA concept of UMTS and allows for re-using these capabilities for any kind of services/applications.

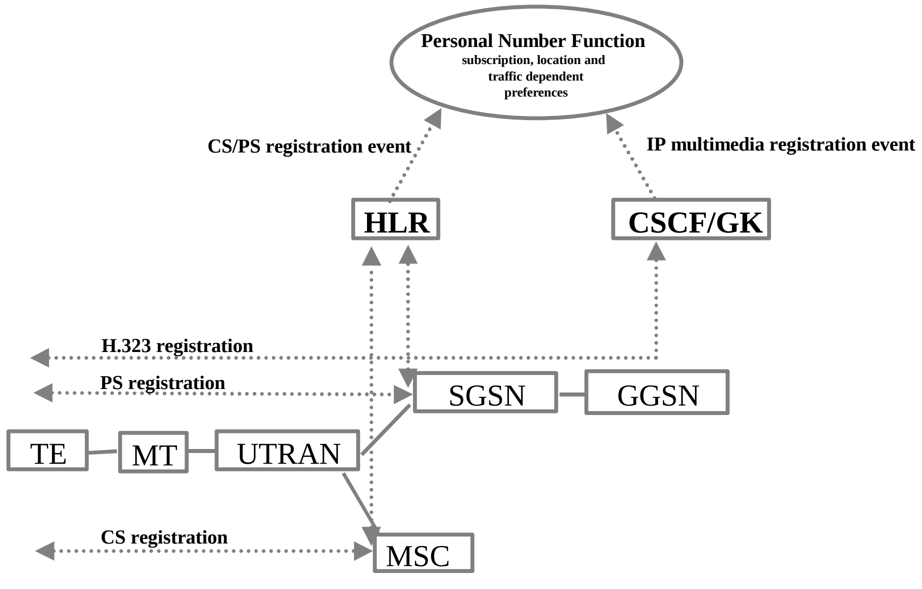
The Personal Interrogation Function may not be identical when interfacing the GSMisdn Domain and the IP MultiMedia domain due to the very different environment, these details are for further study.

Call handling

In the next following sections a few principle call scenarios are shown to highlight how packet and circuit calls interact with respect to the one MSISDN concept. In the examples that follow, H.323 is named just as an IP multimedia example.

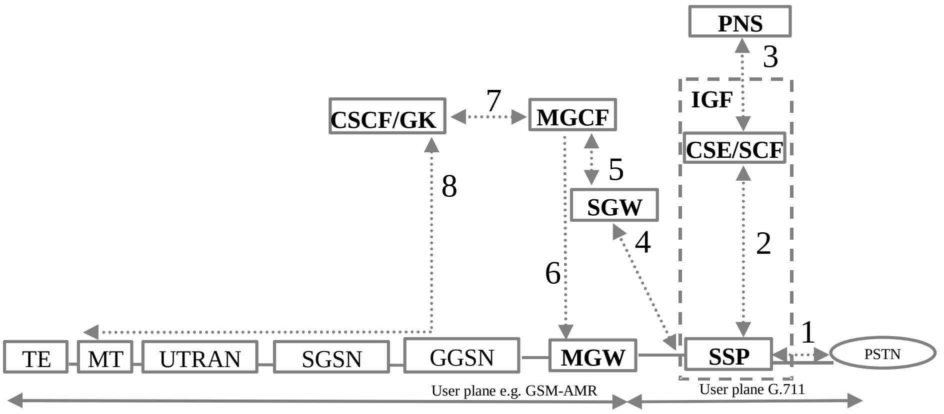
It is here assumed that the home PLMN makes the interrogation towards the Personal Number service, which determines the called user's call reception point preference based on a one MSISDN scheme. The split of CSCF into a visiting and home CSCF has not been considered herein. Optimal routing is not considered either, but remains for further study.

Registration of a CS/PS Mobile Terminal in an R00 Network



Above is shown how a CS/PS mobile terminal e.g. at power on makes three types of registrations. One is a circuit registration, one is a GPRS registration, and one is an IP multimedia e.g. H.323 registration. The latter is done within the GPRS user plane. The HLR indicates to the Personal Number Function that the user is available for CS speech calls. The CSCF indicates to the Personal Number Function that the user is available for IP multimedia voice calls.

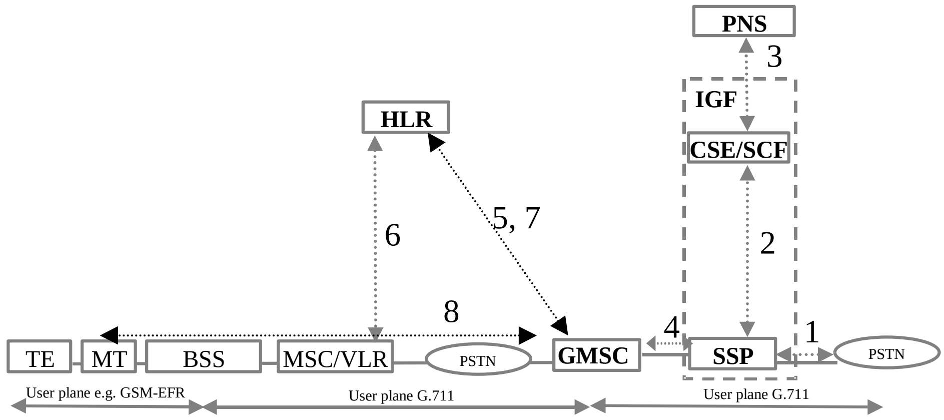
PSTN -> Mobile Subscriber IP Multimedia Voice Call



Above figure shows an incoming call from PSTN to a mobile subscriber who wants telephony calls as IP multimedia voice calls. The SSP together with the CSE/SCE functions makes up the IGF as described above in chapter 1. Of course the SSP can be realised as an SSF within some other network element.

1) Incoming call from PSTN is received in the SSP
2) SSP informs CSE/SCF
3) CSE/SCF interrogates the personal number server for routing instructions based on incomming MSISDN.
4) A routing number is returned (MSISDN with IP multimedia prefix) in SSP which forwards the call to the SGW
5) SGW translates ISUP/STM to ISUP/IP towards MGCF
6) MGCF allocates resources from the MGW
7) MGCF translates the ISUP signalling to H.323 towards CSCF/GK
8) CSCF/GK checks the user's service profile and thereafter routes the IP MM call over GPRS onto the terminal.

PSTN -> Mobile Subscriber Roaming in a GSM Only Network



Above figure shows an incoming speech call from PSTN to a mobile subscriber who is registered under a GSM only PLMN.

1) Incoming call from PSTN is received in the SSP

2) SSP informs CSE/SCF

3) CSE/SCF interrogates the personal number server for routing instructions based on MSISDN

4) A routing number (MSISDN with CS prefix) is returned in SSP which forwards the call to the GMSC

5) GMSC requests routing information from HLR

6) HLR requests roaming number from VLR

7) HLR returns a roaming number to GMSC

8) GMSC forwards the call via PSTN to the destination NW

~~6 Interoperability between GSM and UMTS~~

- ~~Transparency [from a users perspective] of roaming and handover~~
- ~~Re-use of existing subscription profiles~~

~~Note: This list is not exhaustive and is FFS.~~

~~This allows easier management and deployment of a new UMTS network.~~

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~~UMTS is a system supporting handovers between GSM and UMTS in both directions. To support these handovers effectively, the following is required from a dual mode MS/UE supporting simultaneous ISDN/PSTN and packet service in GSM/UMTS:~~

~~Depending upon the solution adopted for GSM-UMTS handover, the MS/UE supporting simultaneous ISDN/PSTN and packet service may be required to perform appropriate update into CN depending on the activity of the UE once the handover between GSM and UMTS is completed. This update is needed to avoid any severe interruptions on the accessibility of packet services after the handover.~~

~~The nature of the update to be made after the handover in both direction, i.e., from GSM to UMTS and from UMTS to GSM, from MS/UE depends on the activity of the UE in the following way:~~

- ~~ISDN/PSTN connection: RA update only (if RA is changed)~~
- ~~Packet connection: LA and RA update (if RA and LA are changed)~~
- ~~Both ISDN/PSTN and packet connection: RA update only (if RA is changed)~~

~~If the RA, LA or both LA and RA are not changed the MS/UE behaviour is for further study~~

~~6.1 Circuit Switched Handover and Roaming Principles~~

~~Introduction of a UMTS Core Network necessitates the inter-connection with legacy systems to allow inter-PLMN roaming and handover.~~

~~For ease of convergence with the existing networks and the introduction of dual mode handsets, roaming and handover to/from UMTS should be performed in the simplest manner that requires as little change as possible to the legacy networks and standards, i.e. inter-MSC handover functionality.~~

~~These principles provide - from a user perspective - transparency of handover and roaming. In addition, operators providing UMTS services should also allow access to legacy networks using existing subscriber profiles and network interfaces.~~

~~Illustrated in Figure 17 shows the introduction of a UMTS Core Network for UMTS phase 1 network configuration. Notice that it leaves the current GSM specifications mainly untouched whereupon the UMTS core network acts towards the GSM MSC like a GSM MSC by providing for example MAP/E for handover purposes. Further, it should be observed that GSM subscriptions belong to the HLR whilst UMTS subscriptions exist in the HLR release 99..~~

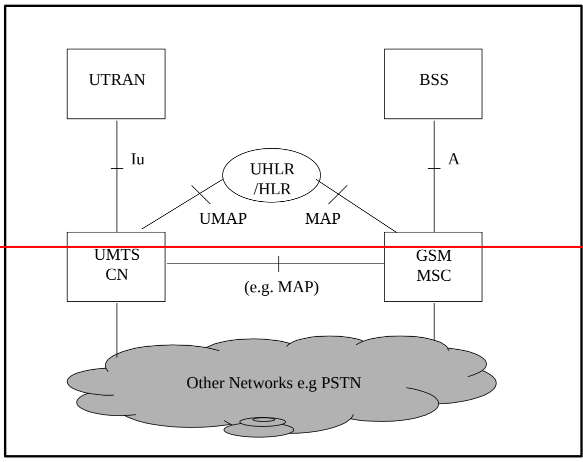

Figure xx. Inter-Operability between GSM and UMTS

~~**Figure 20 Inter-Operability between GSM and UMTS**~~

~~Note: No physical implementation should be taken from the figure. As a further note, no interworking functions are shown to ease clarity, but however should not be precluded.~~

~~From Figure 17 it can be seen that the information exchanged over the Iu must provide the necessary parameters to enable the core networks to communicate via for example the MAP interface for handover purposes.~~

~~Also note that from the above diagram, existing interfaces are used towards the HLR to allow for subscription management based on today's principles using the already defined user profile, providing seamless roaming between the 2nd generation system and UMTS.~~

~~The existing GSM handover procedures should be re-used to minimise the effects on existing GSM equipment (figure 1).~~

- ~~The anchor concept in GSM for inter-MSC handover should be used for inter-system handover between UMTS and GSM.~~

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- ~~The signalling over the A-interface and over the MAP/E-interface should be the same as in GSM phase 2+ with possibly addition of some new or updated information elements in some messages.~~

- ~~For the set up of the handover leg (user plane) standard ISUP/POTS should be used in line with the principles used in GSM.~~

- ~~The control signalling over the Iu-interface at handover between UMTS and GSM should be based on the A-interface signalling at inter-MSC handover in GSM.~~

- ~~The signalling over the Iu-interface at call set up to/from a dual mode UMTS/GSM mobile station, shall include GSM information elements needed for handover from UMTS to GSM.~~
  ~~In the corresponding way the signalling over the A-interface at call set up to/from a dual mode UMTS/GSM mobile shall include UMTS elements needed for handover from GSM to UMTS.~~
  ~~The data are needed to initiate the handover towards the new BSS/RNC.~~

- ~~A target cell based on CGI is sent to the MSC from UTRAN at handover from UMTS to GSM. The CGI points out the target MSC and target BSC.~~
  ~~The target "cell" identifier for UMTS at handover in the direction GSM to UMTS is for further study.~~

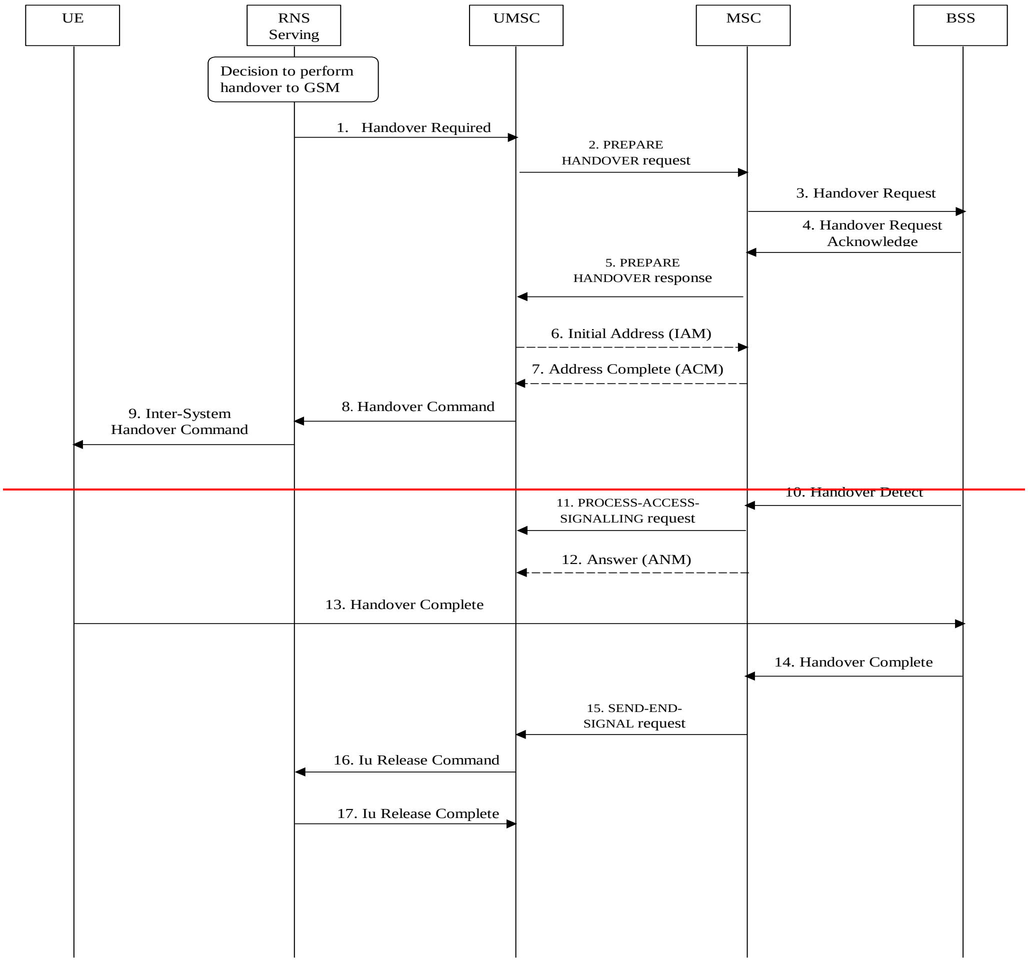
~~6.1.1 UMTS to GSM handover for circuit switched services~~

~~The signalling sequence in figure 1 shows the case when the UMTS MSC (UMSC) and the GSM MSC are located in separate "physical" nodes.~~

~~If the UMSC and MSC are located within the same "physical" node, no MAP signalling and no ISUP signalling are needed between UMSC and MSC.~~

~~For release 99 it is expected that the codec is placed in the anchor or non-anchor UMSC (for the UE in UMTS mode), which will have no impact on the signalling.~~

~~Note: Handling of the user plane is FFS.~~



~~Figure 21. UMTS to GSM handover for circuit switched services e.g. voice~~

1) ~~SRNS initiates the preparation of UMTS to GSM Handover by sending the RANAP message Handover Required to UMSC. This message includes parameters such as Target cell identification and Serving cell identification, both in the form of CGI according to GSM.~~

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~~2) UMSC requests MSC to prepare for UMTS to GSM Handover, by sending the MAP message PREPARE HANDOVER request. The message contains a BSSMAP message Handover Request, to be sent from MSC to BSS. It includes data such as Target and Serving CGI received from the Handover Required message, and data stored in UMSC indicating type of radio resources required.~~

~~3) MSC sends the BSSMAP message Handover Request to BSS which then allocates necessary radio resources in BSS.~~

~~4) When BSS has allocated necessary radio resources it sends the BSSMAP message Handover Request Acknowledge. This message contains all radio-related information that the UE needs for handover, i.e. a complete GSM Handover Command message to be sent transparently via MSC, UMSC, and SRNS to UE.~~

~~5) MSC acknowledges handover preparation by sending the MAP message Prepare Handover Respons to UMSC, including a complete GSM Handover Command message.~~

~~6) UMSC sends the ISUP message IAM to MSC to establish a circuit ISUP connection between UMSC and MSC.~~

~~7) As acknowledgement to IAM, MSC sends the ISUP message ACM back to UMSC.~~

~~8) UMSC sends the RANAP message Handover Command to SRNS, including a complete GSM Handover Command message to be sent to UE.~~

~~9) SRNS sends the RRC message Inter-System Handover Command to UE, including a complete GSM handover Command message, to order the UE to start the execution of handover.~~

~~10) Upon detection of UE in BSS, (by reception of the Layer1 GSM message Handover Access from the UE), which indicates that the correct UE has successful accessed the radio resource in the target GSM-cell, the BSSMAP message Handover Detect is sent from BSS to MSC. MSC may use this condition to switch the connection to the BSS.~~

~~11) MSC sends the MAP message PROCESS-ACCESS-SIGNALLING request to UMSC, including the BSSMAP message Handover Detect. UMSC may use this message as trigger point for switch of the connection to the MSC.~~

~~12) To complete the ISUP signalling the ISUP message ANM is sent from MSC to UMSC.~~

~~13) After Layer 1 and 2 connections are successfully established, the UE sends the GSM message Handover Complete to BSS.~~

~~14) After completed handover, BSS sends the BSSMAP message Handover Complete to MSC.~~

~~15) MSC sends the MAP message SEND-END-SIGNAL request to UMSC, including the BSSMAP message Handover Complete.~~

~~16) UMSC initiates release of resources allocated by the former SRNS.~~

~~17) SRNS acknowledges release of resources.~~

~~6.2 Packet Switched Handover and Roaming Principles~~

~~The introduction of a UMTS core Network as described in section 11.1 illustrates the requirement for inter-connection with the legacy GSM system to allow inter-PLMN roaming and handover.~~

~~Even though there is no current GPRS deployment, the operator may decide to deploy a GPRS network prior to the deployment of a UMTS network. Therefore, the introduction of a UMTS Core Network may require to be inter-connected to the legacy packet network.~~

~~As in the circuit switched case, roaming and handover to/from UMTS should be performed in the simplest manner that requires as little change as possible to the GPRS network and standards, i.e. inter-GSN handover functionality. In addition, access is provided to the GPRS network using the existing subscriber profiles and current network interfaces.~~

~~A similar figure to Figure 17 is illustrated in Figure 19. Notice that it also leaves the current GPRS specifications mainly untouched whereupon the UMTS core network acts towards the GSN like a GSN by providing for example Gn. Further, it should be observed that GPRS subscriptions belong to the HLR whilst UMTS subscriptions exist in the HLR release 99.~~

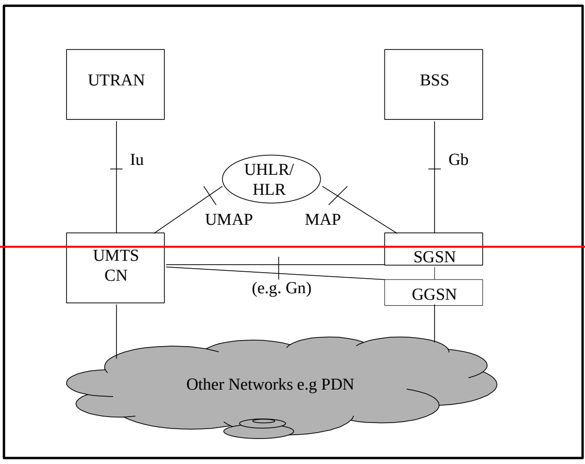

Figure xx. Inter-Operability between GSNs and UMTS

~~Figure 22 Inter-Operability between GSNs and UMTS~~

~~Note: No physical implementation should be taken from Figure 19. As a further note, no interworking functions are shown to ease clarity, but however should not be precluded.~~

~~From Figure 19 it can be seen that to provide inter-working between legacy packet switched and UMTS packet switched services, the information exchanged over the Iu must provide the necessary parameters to enable the core networks to communicate via for example the Gn interface for handover purposes.~~

~~Also note that from the above diagram, the same principles are used as in the circuit switched services to provide seamless roaming.~~

~~6.2.1 Implications~~

- ~~The active PDP context resides in the same GGSN even after a handover between GSM and UMTS (both directions). This corresponds in principle to the anchor concept on the circuit switched side, but note that whereas packet sessions are long lived, the anchor MSC remains only for the duration of a CS call (typically much shorter than a packet session).~~

- ~~Assuming an internal structure in UMTS CN that contains logical GGSN and SGSN nodes, the signalling over the inter-system GGSN-SGSN interface should be a joint evolution of Gn for the GSM system and UMTS. I.e., when Gn evolves in the sequence of GSM releases, Gn should include any new or updated information necessary for interoperation.~~

- ~~The corresponding SGSN-SGSN inter-system interface (also Gn) should also be evolved together. However, in this case the changes relative to the current GPRS release may possibly be more profound.~~

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~~6.2.2 Signalling procedures~~

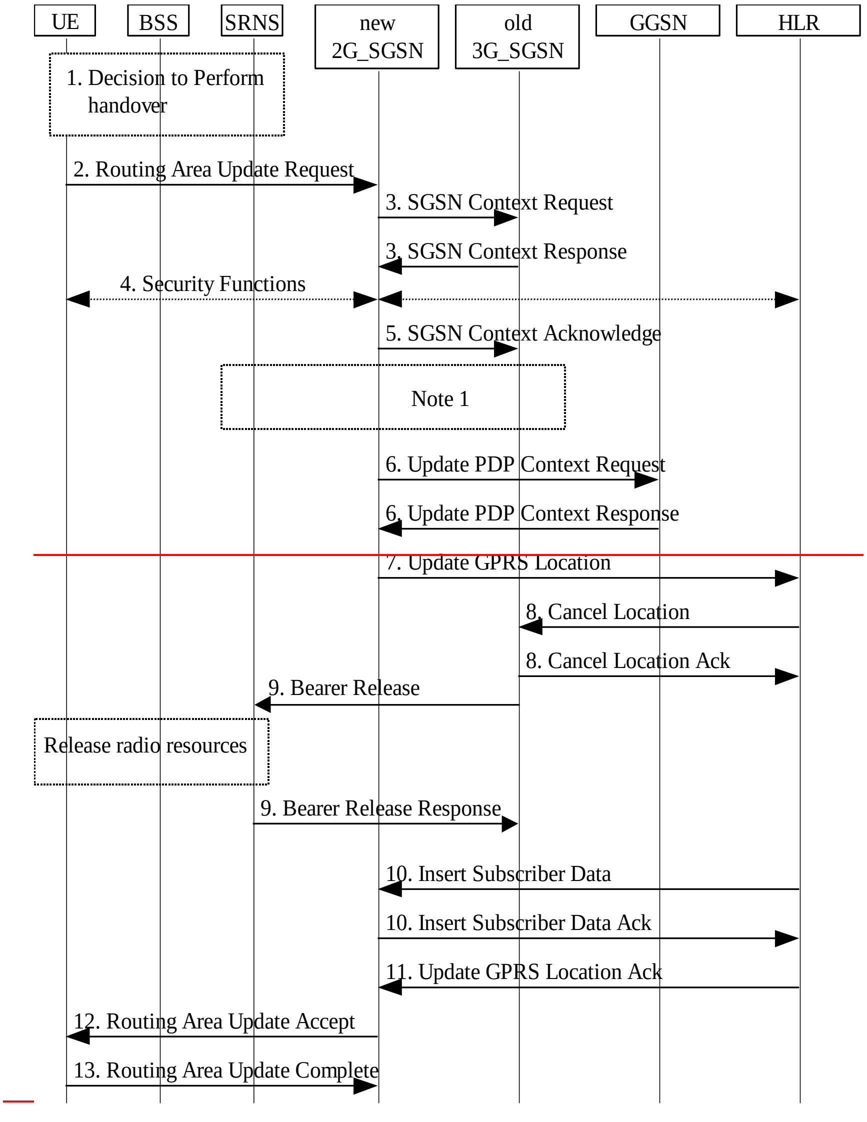
~~The signalling procedures shows how handover UMTS <-> GSM GPRS can be done. The parameters carried by each message is not complete and shall be seen as examples of important information carried be the messages.~~

~~The signalling sequences shows the case when the UMTS 3G_SGSN and the GPRS 2G_SGSN are located in separate "physical" nodes.~~

~~If the 3G_SGSN and 2G_SGSN are located within the same "physical" node, no signalling are needed between 3G_SGSN and 2G_SGSN.~~

~~For handover in the UMTS to GSM GPRS direction the intention is to re-use the handover principles of GSM GPRS today in order to limit the changes in GSM GPRS and to take the changes if any on the UMTS side. The below specified messages is standard GSM 2+ messages (when applicable)~~

~~Handover from UMTS to GSM GPRS~~



~~Figure 23: UMTS to GSM GPRS, Inter SGSN Routing Area Update Procedure~~

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~~1) The UE [2] or UTRAN [2] decides to perform handover which leads to that the UE switch to the new cell under the new system.~~

~~2) The UE sends a Routing Area Update Request (old RAI, old P-TMSI) to the new 2G_SGSN. The BSS shall add the Cell Global Identity including the RAC and LAC of the cell where the message was received before passing the message to the 2G_SGSN.~~

~~3) The new 2G_SGSN sends SGSN Context Request (old RAI, old P-TMSI, New SGSN Address) to the old 3G_SGSN to get the MM and PDP contexts for the UE (The old RAI received from the UE is used to derive the old 3G_SGSN address). The old 3G_SGSN responds with SGSN Context Response (MM Context, e.g. IMSI, PDP Contexts, e.g. APN).~~

~~4) Security functions may be executed.~~

~~5) The new 2G_SGSN sends an SGSN Context Acknowledge message to the old 3G_SGSN. This informs the old 3G_SGSN that the new 2G_SGSN is ready to receive data packets belonging to the activated PDP contexts.~~

~~6) The new 2G_SGSN sends Update PDP Context Request (new SGSN Address) to the GGSN concerned. The GGSN update their PDP context fields and return Update PDP Context Response.~~

~~7) The new 2G_SGSN informs the HLR of the change of SGSN by sending Update GPRS Location (SGSN Number, SGSN Address, IMSI) to the HLR.~~

~~8) The HLR sends Cancel Location (IMSI) to the old 3G_SGSN. The old 3G_SGSN removes the MM and PDP contexts.~~
~~The old 3G_SGSN acknowledges with Cancel Location Ack (IMSI).~~

~~9) The old 3G_SGSN request the SRNS to release the radio resources by sending Bearer Release. The SRNS responds with Bearer Release Response.~~

~~10) The HLR sends Insert Subscriber Data (IMSI, GPRS subscription data) to the new 2G_SGSN. The 2G_SGSN constructs an MM context for the UE and returns an Insert Subscriber Data Ack (IMSI) message to the HLR.~~

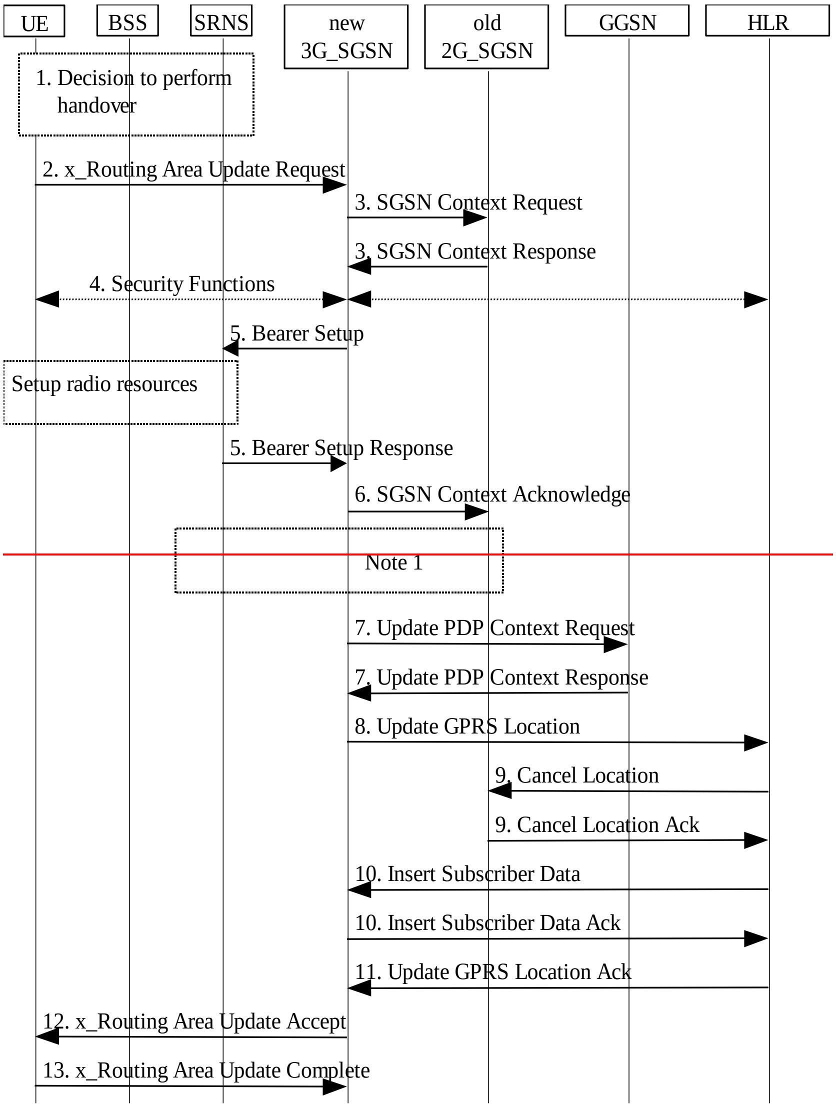
~~11) The HLR acknowledges the Update Location by sending Update GPRS Location Ack (IMSI) to the new 2G_SGSN.~~

~~12) The new 2G_SGSN validates the UE's presence in the new RA. The new 2G_SGSN constructs MM and PDP contexts for the UE. A logical link is established between the new 2G_SGSN and the UE. The new 2G_SGSN responds to the UE with Routeing Area Update Accept (P-TMSI).~~

~~13) The UE acknowledges the new P-TMSI with a Routing Area Update Complete (P-TMSI).~~

~~Note 1: The functionality for forward of packets and handling of GTP sequence numbers is a subject fore more investigation, i.e. FFS.~~

~~Handover from GSM GPRS to UMTS~~



~~Figure 24: GSM GPRS to UMTS, Inter SGSN Routing Area Update Procedure~~

1) ~~The UE/network decides to perform handover which leads to that the UE switch to the new cell, details for this is FFS.~~

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2) ~~The UE sends a x_Routing Area Update Request (old RAI, old P-TMSI) to the new 3G_SGSN. The SRNS shall add an identifier of the area where the message was received before passing the message to the 3G_SGSN.~~

3) ~~The new 3G_SGSN sends SGSN Context Request (old RAI, old P-TMSI, New SGSN Address) to the old 2G_SGSN to get the MM and PDP contexts for the UE (The old RAI received from the UE is used to derive the old 2G_SGSN address). The old 2G_SGSN responds with SGSN Context Response (MM Context, e.g. IMSI, PDP Contexts, e.g. APN).~~

4) ~~Security functions may be executed.~~

5) ~~The new 3G_SGSN request the SRNS to establish of a radio access bearer by sending Bearer Setup to the SRNS. The SRNS responds with Bearer Setup Response.~~

~~6) The new 3G_SGSN sends an SGSN Context Acknowledge message to the old 2G_SGSN. This informs the old 2G_SGSN that the new 3G_SGSN is ready to receive data packets belonging to the activated PDP contexts.~~

~~7) The new 3G_SGSN sends Update PDP Context Request (new SGSN Address) to the GGSN concerned. The GGSN update their PDP context fields and return Update PDP Context Response.~~

~~8) The new 3G_SGSN informs the HLR of the change of SGSN by sending Update GPRS Location (SGSN Number, SGSN Address, IMSI) to the HLR.~~

~~9) The HLR sends Cancel Location (IMSI) to the old 2G_SGSN. The old 2G_SGSN removes the MM and PDP contexts.~~
~~The old 2G_SGSN acknowledges with Cancel Location Ack (IMSI).~~

~~10) The HLR sends Insert Subscriber Data (IMSI, GPRS subscription data) to the new 3G_SGSN. The 3G_SGSN constructs an MM context for the UE and returns an Insert Subscriber Data Ack (IMSI) message to the HLR.~~

~~11) The HLR acknowledges the Update GPRS Location by sending Update Location Ack (IMSI) to the new 3G_SGSN.~~

~~12) The new 3G_SGSN validates the UE's presence in the new RA. The new 3G_SGSN constructs MM and PDP contexts for the UE. A logical link is established between the new SGSN and the UE. The new 3G_SGSN responds to the UE with x_Routing Area Update Accept (P-TMSI).~~

~~13) The UE acknowledges the new P-TMSI with a x_Routing Area Update Complete (P-TMSI).~~

~~Note 1: The functionality for forward of packets and handling of GTP sequence numbers (within the box) is a subject fore more investigation, i.e. FFS.~~

7 Network Migration And Evolution

The installed base of GSM networks will be very comprehensive at the time of the UMTS roll out. These GSM networks will co-operate very closely with and in many cases be partly integrated into the overall UMTS network. Thus network migration and evolution is a very fundamental aspect to consider when standardising UMTS.

7.1 Network Migration Scenarios

A number of principally different network migration scenarios can be envisioned, e.g.:

- GSM to GSM release 99 (GSM operator with no UMTS licence and no UMTS roaming/handover agreements).
- GSM to GSM release 99 with support for dual mode 'UMTS visitors' (GSM operator with no UMTS licence but with UMTS roaming/handover agreements).
- GSM to GSM/UMTS (GSM operator with a UMTS licence).
- UMTS only PLMN (new UMTS operator with GSM roaming/handover agreements). This scenario is more a matter of network 'compatibility' rather than network migration.

A basic assumption is that the provision of UMTS services in most cases will start, from a radio coverage point of view, within 'islands in a sea of GSM BSS'.

7.2 network migration and evolution requirements

1) The UMTS standard shall consider all aspects of network migration and shall describe the migration process from GSM release 98 to UMTS/GSM release 99, including the aspect of partly updated networks and its consequences on end-user services etc.
2) While fulfilling the SMG1 requirements the UMTS standard shall aim at minimising the impact on the existing GSM networks delivering only GSM. It is recognised that GSM/GPRS standards will need developments for UMTS however these should not adversely impact the networks that offer GSM only.
3) It shall be possible to perform the network migration process of a PLMN independently of co-operating PLMNs.
4) It shall be possible to gradually migrate a PLMN, i.e. the UMTS standard shall allow network elements compliant with different GSM releases to co-exist within a PLMN.
5) The impact on end-user service level for partly updated PLMN(s) is FFS.
6) Internetworking within a PLMN as well as between different PLMNs shall allow operators to utilise current backbone networks (dedicated for GSM traffic only or carrying non-mobile traffic as well according to non-PLMN specific standards).
7) A GSM/UMTS mobile terminal shall be reachable from an external network (PSTN/ISDN, IP, X.25) regardless of the mobile terminal being served by BSS or UTRAN.
8) A terminal in an external network, as well as the external networks themselves, shall not need to know if the GSM/UMTS mobile terminal is served by BSS or UTRAN.
9) The user equipment shall not need to change the E.164 or IP address at handover between UTRAN and BSS.

8 Protocol Architecture

8.1 I_U Signalling Bearer Requirements for IP Domain

8.1.1 Connectionless and Connection Oriented Services

Connection-oriented and connection-less I_U Signalling Bearers are required.

8.1.2 Dynamic Bandwidth Allocation

The I_U Signalling Bearer shall support rapid and flexible allocation and de-allocation of I_U transport resources.

8.1.3 Reliable Transfer

The I_U Signalling Bearer shall provide reliable delivery of signalling data.

8.1.4 Flow Control

The I_U Signalling Bearer shall provide throttling mechanisms to adapt to intermittent congestion in the UTRAN or Core Network.

8.1.5 Redundancy and Load Sharing

To handle detected failures and signalling data congestion, the I_U Signalling Bearer shall be capable of dynamically routing over alternate routes that minimise delay. If the delay metrics over alternative routes are identical, the I_U Signalling Bearer shall be capable of spreading traffic over the identical paths, thus performing load sharing.

8.1.6 Large Pdu Size

To support large transactions, it is important for the I_U Signalling Bearer to provide a Signalling Data Unit size, large enough to allow for all signalling messages to be transferred without fragmentation.

8.1.7 Signalling Bearer Management

To support supervision of I_U Signalling Bearers, mechanisms for managing I_U Signalling Bearers shall be used to provide status information to the RANAP for individual UE(s). The signalling bearer shall also maintain a consistent UE Activation State in the access and the core network.

8.1.8 Transport Media Independence

The I_U Signalling Bearer shall be independent of the underlying transport media (e.g. ATM).

9 History

Document history		
V1.0.0	June 1999	Creation of document from 23.20, all sections except 7
V2.0.0	June 1999	Some editorial changes in order to prepare the document for the approval by the TSG SA, June 1999 meeting
V3.0.0	July 1999	Template changed, clauses and sub-clauses numbering corrected, administrative clauses added.
v.3.1.0	October 1999	Incorporation of all the Change Requests approved at TSG SA#5