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Contents

Foreword

This Technical Specification (TS) has been produced by ETSI 3rd Generation Partnership Project (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 the present document, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

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1 Scope

This Technical Specification (TS) describes the Service Principles for PLMNs specified by 3GPP. Principles and requirements for interworking with WLAN are covered in TS 22.234 [35].

3GPP specifications provide integrated personal communications services. The system will support different applications ranging from narrow-band to wide-band communications capability with integrated personal and terminal mobility to meet the user and service requirements of the 21st century.

3GPP specifications allow the realisation of a new generation of mobile communications technology for a world in which personal communications services should allow person-to-person calling, independent of location, the terminal used, the means of transmission (wired or wireless) and the choice of technology. Personal communication services should be based on a combination of fixed and wireless/mobile services to form a seamless end-to-end service for the user.

3GPP specifications should be in compliance with the following objectives:

a) to provide a single integrated system in which the user can access services in an easy to use and uniform way in all environments;

b) to allow differentiation between service offerings of various serving networks and home environments;

c) to provide a wide range of telecommunications services including those provided by fixed networks and requiring user bit rates of up to 100 Mbit/s as well as services special to mobile communications. These services should be supported in residential, public and office environments and in areas of diverse population densities. These services are provided with a quality comparable with that provided by fixed networks such as ISDN and fixed broadband Internet access;

d) to provide services via hand held, portable, vehicular mounted, movable and fixed terminals (including those which normally operate connected to fixed networks), in all environments (in different service environments - residential, private domestic and different radio environments) provided that the terminal has the necessary capabilities;

e) to provide support of roaming users by enabling users to access services provided by their home environment in the same way even when roaming.

f) to provide audio, data, video and particularly multimedia services;

g) to provide for the flexible introduction of telecommunication services;

h) to provide within the residential environment the capability to enable a pedestrian user to access all services normally provided by fixed networks;

i) to provide within the office environment the capability to enable a pedestrian user to access all services normally provided by PBXs and LANs;

j) to provide a substitute for fixed networks in areas of diverse population densities, under conditions approved by the appropriate national or regional regulatory authority.

k) to provide support for interfaces which allow the use of terminals normally connected to fixed networks.

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. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

2.1 Normative references

[1] 3GPP TS 22.105 "Services and Service Capabilities"

[2] Void

[3] 3GPP TS 22.038: "(U)SIM Application Toolkit (USAT); Service description; Stage 1".

[4] 3GPP TS 22.001: "Principles of Circuit telecommunication services supported by a Public Land Mobile Network (PLMN)".

[5] 3GPP TS 22.004: "General on supplementary services"

[6] 3GPP TS 22.030: "Man-Machine Interface (MMI) of the User Equipment (UE)"

[7] 3GPP TS 22.066: "Support of Mobile Number Portability (MNP); Service description; Stage 1"

[8] 3GPP TS 22.079: " Support of Optimal Routeing (SOR); Service definition; Stage 1".

[9] 3GPP TS 22.129: "Handover Requirements between UTRAN and GERAN or other Radio Systems".

[10] 3GPP TS 33.102: "Security Architecture".

[11] 3GPP TS 22.011: "Service Accessibility".

[12] 3GPP TS 22.016: "International mobile Station Equipment Identities (IMEI)".

[13] 3GPP TS 24.008: "Mobile Radio Interface Layer 3 Specification".

[14] 3GPP TS 22.003: "Circuit Teleservices supported by a Public Land Mobile Network (PLMN)".

[15] 3GPP TS 21.133: "Security Threats and Requirements".

[16] 3GPP TS 33.120: "Security Principles".

[17] 3GPP TS 22.042: "Network Identity and Time Zone, Service Description, Stage 1".

[18] 3GPP TS 42.009: "Security Aspects".

[19] 3GPP TS 31.102: "USIM Application Characteristics".

[20] 3GPP TS 23.221 "Architectural Requirements".

[21] 3GPP TS 22.002: "Circuit Bearer Services (BS) supported by a Public Land Mobile Network (PLMN)".

[22] 3GPP TS 22.060: "General Packet Radio Service (GPRS) ; Service description; Stage 1".

[23] 3GPP TS 29.002: "Mobile Application Part (MAP) specification".

[24] 3GPP TR 23.972: "Circuit switched multimedia telephony".

[25] 3GPP TS 22.140: " Multimedia Messaging Service (MMS); Stage 1".

[26] 3GPP TS 22.226: "Global Text Telephony, Stage 1".

[27] 3GPP TS 22.228: " Service requirements for the Internet Protocol (IP) multimedia core network subsystem (IMS); Stage 1".

[28] RFC 3261: "SIP: Session Initiation Protocol".

[29] 3GPP TR 21.905: " Vocabulary for 3GPP Specifications".

[30] 3GPP TS 26.233: "Packet Switched Streaming Service (PSS) ; General Description".

[31] 3GPP TS 26.234: "Packet Switched Streaming Service (PSS) ; Protocols and Codecs".

[32] 3GPP TR 22.934: "Feasibility study on 3GPP system to Wireless LAN (WLAN) interworking".

[33] RFC 2486: "The Network Access Identifier".

[34] 3GPP TS 51.011: "Specification of the Subscriber Identity Module - Mobile Equipment (SIM-ME) interface Release 4)".

[35] 3GPP TS 22.234: "Requirements on 3GPP system to wireless local area network (WLAN) interworking".

[36] 3GPP TS 31.101: "UICC-terminal interface; Physical and logical characteristics".

[37] OMA Device Management V1.2 specifications

[38] OMA Client Provisioning V1.1 specifications

[39] void

[40] 3GPP TS 22.173: " IP Multimedia Core Network Subsystem (IMS) Multimedia Telephony Service and supplementary services; Stage 1".

[41] 3GPP TS 22.082: "Call Forwarding (CF) supplementary services - Stage 1".

[42] 3GPP TS 22.278: "Service Requirements for the Evolved Packet System (EPS)".

[44] 3GPP TS 22.071: "Location Services (LCS); Service description; Stage 1".

[45] 3GPP TR 22.985: "Service requirement for the 3GPP User Data Convergence (UDC), Release 9".

[46] DD CEN/TS 15722:2009 "Road Transport and Traffic Telematics, eSafety, eCall Minimum Set of Data (MSD)"

[47] 3GPP TS 23.226: "Global text telephony (GTT); Stage 2"

[48] 3GPP TS 22.220: " Service requirements for Home Node B (HNB) and Home eNode B (HeNB) ".

[49] ETSI TS 181 019 V2.0.0 (2007-11): "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Business Communication Requirements".

[50] 3GPP TS 23.335: "User Data Convergence (UDC); Technical realization and information flows; stage 2".

[51] OpenID Foundation: "OpenID Authentication 2.0", http://openid.net/specs/openid-authentication-2_0.html.

[52] 3GPP TS 22.368: "Service requirements for Machine-Type Communications (MTC); Stage 1".

[53] OMA Presence API: "OMA-TS-REST_NetAPI_Presence-V1_0-20130212-C".

[54] IETF RFC-5491: "GEOPRIV Presence Information Data Format Location Object (PIDF-LO) Usage Clarification, Considerations, and Recommendations".

[55] IETF RFC-5139: "Revised Civic Location Format for Presence Information Data Format Location Object (PIDF-LO)".

[56] 3GPP TS 23.032: “Universal Geographical Area Description (GAD)”

2.2 Informative references

[43] GSMA PRD IR.34: "Inter-Service Provider IP Backbone Guidelines"

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in TR 21.905 [1] and the following apply. A term defined in the present document takes precedence over the definition of the same term, if any, in TR 21.905 [1].

3GPP SSO Authentication: Authentication performed between an SSO-capable UE and 3GPP SSO Identity Provider using Operator-controlled credentials and without requiring user involvement.

3GPP SSO Identity Provider: An entity that maintains Operator-controlled identity and credential information for a user, performs 3GPP SSO Authentication, and asserts the user's identity to a Data Application Provider.

3rd Party SSO Identity Provider: An entity that maintains identity and credential information (that is not Operator-controlled) for a user, performs authentication, and asserts the user's identity to a Data Application Provider.

Attended Data Traffic: Data traffic of which the user is aware he/she initiated, e.g. based on the screen/keypad lock being deactivated, length of time since the UE last received any input from the user, known type of application (e.g. an application monitoring a user's health – "mHealth" – which may need its data always treated as Attended Data Traffic.)

eCall: A manually or automatically initiated emergency call,(TS12) from a vehicle, supplemented with a minimum set of emergency related data (MSD), as defined under the EU Commission's eSafety initiative.

Data Application Provider: An entity that offers data application services to users (e.g., over the public Internet). The data applications can be browser or non-browser based services.

E-UTRAN Sharing: The sharing of E-UTRAN among a number of operators.

GERAN or UTRAN Sharing: The sharing of GERAN or UTRAN among a number of operators.

Hosting E-UTRAN Operator: The Operator that has operational control of a Shared E-UTRAN. With regard to management of the Shared E-UTRAN the Hosting E-UTRAN Operator is a Master Operator [29].

Hosting RAN: The Shared RAN that is owned or controlled by the Hosting RAN Operator.

Hosting RAN Operator: The Operator that has operational control of a Shared E-UTRAN, UTRAN or GERAN

IMS Centralized Services: The provision of communication services wherein services and service control are based on IMS mechanisms and enablers, and support is provided for a diversity of access networks (including CS domain and IP based, wireless and wireline), and for service continuity between access networks.

MSD: The Minimum Set of Data [46] forming the data component of an eCall sent from a vehicle to a Public Safety Answering Point or other designated emergency call centre. The MSD has a maximum size of 140 bytes and includes, for example, vehicle identity, location information and time-stamp.

Participating Operator: Authorized operator that is sharing E-UTRAN, UTRAN or GERAN resources provided by a Hosting RAN Operator

RAN user plane congestion: The situation where the demand for RAN resources to transfer user data exceeds the available RAN capacity to deliver the user data for a significant period of time in the order of few seconds or longer.

(S)Gi-LAN: The network infrastructure connecting to 3GPP network over the SGi or Gi reference point that provides various IP-based services (e.g. NAT, antimalware, parental control, DDoS protection, video optimization).

Shared E-UTRAN: E-UTRAN that is shared among a number of operators.

Shared RAN: GERAN, UTRAN or E-UTRAN that is shared among a number of operators.

Shared GERAN or UTRAN: GERAN or UTRAN that is shared among a number of operators.

SSO Provider: An entity that provides SSO Service. The SSO Provider enables a user to authenticate to an IdP and thereby to have their identity asserted to a DAP. Each data application, whether provided by different DAPs or the same DAP, may have its own policy regarding authentication. In the 3GPP SSO Service, the SSO Provider is the 3GPP Operator.

SSO Service: A service in which the user of a data application is authenticated once, and as a result of that authentication is provided with seamless and transparent access to multiple data applications offered by one or more Data Application Providers.

SSO Local User Authentication: Authentication performed by the UE that establishes the presence of the registered user of the data application by requiring input which only the registered user would be able to provide.

Unattended Data Traffic: Data traffic of which the user is unaware he/she initiated, e.g. based on the screen/keypad lock being activated, length of time since the UE last received any input from the user, known type of app (e.g. an application monitoring a user's health – "mHealth" – may need its data never treated as Unattended Data Traffic.)

Further definitions are given in 3GPP TR 21.905 [29].

3.2 Abbreviations

For the purposes of the present document, the abbreviations given in TR 21.905 [1] and the following apply.
An abbreviation defined in the present document takes precedence over the definition of the same abbreviation, if any, in TR 21.905 [1].

DAP	Data Application Provider
IdP	Identity Provider
IVS	In Vehicle System (eCall terminal and associated sub-systems in vehicle)
ME	Mobile Equipment
OTT	Over The Top
PC	Personal Computer
SSO	Single Sign-On

4 General

4.1 Aims of 3GPP specifications

It shall be capable of delivering audio, text, video and graphics direct to people and provide them with access to the next generation of information based services. It moves mobile and personal communications forward from existing systems, delivering mass market low-cost digital telecommunication and IP-based multimedia services.

The aims are:

- to enable users to access a wide range of telecommunications services, including many that are today undefined as well as multi-media and high data rates.
- to facilitate the provision of a high quality of service (particularly speech quality) similar to that provided by fixed networks;
- to facilitate the provision of small, easy to use, low cost terminals with long talk time and long standby operation;
- to provide an efficient means of using network resources (particularly radio spectrum).

4.2 Standardisation of Service Capabilities

Existing systems have traditionally standardised the complete sets of teleservices, applications and supplementary services which they provide. As a consequence, substantial efforts are often required to introduce new services or simply to modify the existing one (customisation). This makes it more difficult for operators to differentiate their services. At the same time however, this may reduce the complexity of providing a service across different operators' networks.

3GPP shall therefore preferentially standardise service capabilities. In circumstances where the service is meant to be used across different operators' networks, hence a common specification set is of paramount importance, the service should be standardised to a level of detail sufficient to ensure interoperability and interworking across different operators' networks. Service capabilities consist of bearers defined by QoS parameters and the mechanisms needed to realise services. These mechanisms include the functionality provided by various network elements, the communication between them and the storage of associated data. This TS provides a conceptual description of a service architecture and architecture requirements which aim to provide service capabilities. It is intended that these standardised capabilities should provide a defined platform which will enable the support of speech, video, multi-media, messaging, data, teleservices, user applications and supplementary services and enable the market for services to be determined by users and home environments.

4.2.1 Provision of service capabilities in shared networks

The provision of services and service capabilities that is possible to offer in a network shall not be restricted by the existence of the network sharing It shall be possible for a core network operator to differentiate its service offering from other core network operators within the shared network.

It shall be possible to control the access to service capabilities offered by a shared network according to the core network operator the user is subscribed to.

4.3 Efficient Use of Network Resources

4.3.1 Network Traffic Patterns

Service capabilities shall take account of the discontinuous and asymmetric nature of most teleservices, multimedia services and user applications and consider the overheads and signalling surge caused by frequent transmissions of small amount of data by mobile data application, in order to make efficient use of network resources (particularly radio resources).

4.3.2 Mass Simultaneous Registration

When a large number of subscribers enter in a registration area in which they have not registered, the core and radio access network shall be able to provide a capability to optimize the mass simultaneous registration traffic at a given instance of time. The core and radio access network shall be able to keep providing the service (e.g. mobile originated and mobile terminated services) without interruption for those subscribers who are originally in the cell which receive the mass simultaneous registration traffic.

4.3.3 Radio Interface

Service capabilities shall be provided in a wide range of radio operating environments (where a radio environment is characterised in terms of propagation environment, mobile equipment relative speeds and traffic characteristics). Although 3GPP aims to minimise the number of radio interfaces and to maximise commonality between them, it may utilise several radio interfaces, each optimised for different environments. Each radio interface may provide differing service capabilities. 3GPP specifications include UTRA(N) radio interface supporting two modes (TDD and FDD), an Evolved UTRA(N) radio interface and GERAN radio interface. Additionally, it may be possible to connect to the 3GPP system using radio interfaces and fixed access technologies specified outside of 3GPP.

3GPP specifications shall provide a mechanism which will enable a piece of user equipment (UE) to adapt to different radio interfaces as necessary and to determine the service capabilities available. The specifications shall also provide a mechanism which will enable a UE to select radio interfaces capable of providing appropriate service capabilities and support mobility between multiple radio interfaces.

4.3.4 Real-time Resource Usage

To enable network operators to render services efficiently, dimension their networks and set tariffs that more accurately reflect radio resource usage, real time information on resource usage is needed. When requested, it shall be possible for the serving cell type (e.g. RAT), cell ID / UTRAN Service Area Identity and cell / Service Area capability usage (e.g. HSDPA, E-DCH) information to be made available to the core network. Cell / Service Area capability usage information may include, for example, user(s) identity, start and finish time of data transfer, up-link and down-link data rates, volumes of data and other statistical information.

4.3.5 Selected IP Traffic Offload (SIPTO) for PS Domain only

4.3.5.1 Common Requirements for SIPTO in the Mobile Operator Network and SIPTO at the Local Network

The 3GPP system shall be able to offload selected traffic (e.g. internet) towards a defined IP network close to the UE's point of attachment to the access network.

The following requirements apply to Selected IP Traffic Offload:

- The mobile operator may enable/disable Selected IP Traffic Offload on a per UE per defined IP network basis (e.g. based on tariff, subscription type etc.).
- It shall be possible for IP traffic of a UE associated with a particular defined IP network to be offloaded while IP traffic of that same UE associated with other defined IP network(s) is not offloaded.
- It shall be possible to perform Selected IP Traffic Offload for pre-Release 10 UEs.
- Offloading selected IP traffic for a UE shall not affect services running in parallel for the same UE.
- The mobile operator shall be able to collect signalling performance measurements (e.g. session connection/disconnection, etc) related to Selected IP Traffic Offload for each user.
- Selected IP Traffic Offload shall not compromise the security of the mobile operator's network.
- Service Continuity of IP data session(s) for Selected IP Traffic Offload may be supported during the following mobility events:
    - mobility between the macro network and H(e)NBs; and

- mobility between H(e)NBs.

  During both these mobility events, based on home mobile operator policies, the impact of mobility events as perceived by the user shall be reduced by minimising any interruption to the data flow.

- It shall be possible for the HPLMN to provide the VPLMN with the following information for a particular user:
  - An indication of whether the user's IP traffic is permitted to be subjected to Selected IP Traffic Offload in the visited network;
  - The defined IP network(s) for which Selected IP Traffic Offload is permitted.

Requirements specific to SIPTO at the local residential/enterprise IP network can be found in section 5.9 in [48].

Some types of services (e.g. streaming services, VOIP, VPN, HTTPS-Based Services) cannot tolerate a change of IP address of the UE without disruption of the service.

SIPTO can be performed with or without coordination between the UE and the network. The following requirements apply to coordinated SIPTO:

- The 3GPP system shall be able to support multiple connections that are associated with the same defined IP network where each connection may or may not support IP address preservation.
- The 3GPP system shall be able to determine if an IP flow requires IP address preservation or not. Based on this determination, the 3GPP network shall be able to offload selected IP traffic in coordinated manner between UE and the network, in order to minimize service disruption.
- The 3GPP system shall be able to detect when a connection becomes suboptimal and decide when to establish a new optimal connection to the same defined IP network or use an existing connection.

Note 1: The definition of optimal and suboptimal can be based on a number of implementation criteria like geography, topology and load balancing etc.

- The 3GPP system shall minimize the number of connections of a UE without disrupting the UE’s services, e.g. to ensure economical use of network resources.
- The 3GPP system shall be able to ensure that the actual average aggregate bit rate for IP flows of packet data network connections associated with the same packet data network does not significantly exceed the subscribed aggregate maximum bit rate for this packet data network when two connections are used with the same defined IP network.

Note 2: Requirements for Coordinated SIPTO do not apply to IMS.

4.3.5.2 Requirements for SIPTO in the Mobile Operator Network

The following requirements apply to Selected IP Traffic Offload in the mobile operator network:

- The mobile operator shall be able to enable/disable Selected IP Traffic Offload for certain parts of the network.
- Selected IP Traffic Offload shall not compromise integrity and confidentiality of offloaded traffic.
- The mobile operator shall be able to collect statistics for the offloaded traffic for each user.
- The network shall be able to perform Selected IP Traffic Offload without any user interaction based on mobile operator policies.
- Service Continuity of IP data session(s) within the mobile operator network shall be supported for Selected IP Traffic Offload. Based on home mobile operator policies, the impact of mobility events within the macro network as perceived by the user shall be reduced by either:
  - minimising any interruption to the data flow; or
  - preventing interruption to the data flow e.g. for voice services.
- Service Continuity of IP data session(s) for Selected IP Traffic Offload may be supported during the following mobility events:

- mobility between the macro network and H(e)NBs; and
- mobility between H(e)NBs.

During both these mobility events, based on home mobile operator policies, the impact of mobility events as perceived by the user shall be reduced by preventing interruption to the data flow e.g. for voice services.

4.4 Compatibility with Global Standards

3GPP specifications aim to be compatible with IMT-2000 and to provide global terminal mobility (roaming), enabling the user to take his/her terminal to different regions of the world and to be provided with services. It is probable that different regions of the world will adopt different radio interface technologies. IMT-2000, as a global standard, should therefore enable a IMT-2000 terminal to determine the radio interface technology and the radio interface standard used in a region. Global terminal roaming also requires the global standardisation of service capabilities. As far as possible the method of indication of the radio interface standard and available service capabilities shall be aligned with IMT-2000.

3GPP specifications shall enable users to access the services provided by their home environment in the same way via any serving network provided the necessary service capabilities are available in the serving network.

The 3GPP specifications will be available for the partner organisations to adopt as their regional standards. For example in Europe, ETSI may adopt them as standards for both GSM and UMTS.

4.5 Void

4.6 Functionality of Serving Network and Home Environment

The following functionality shall be the responsibility of the home environment:

- User Authentication.
- SIM/USIM Issue.
- Billing.
- User Profile/VHE Management.

The following functionality shall be the responsibility of the serving network:

- Radio or other means of access.
- Transport and signalling.

The following functionality may be the responsibility of either the serving network, the home environment or an appropriate combination of both

- Service Control.
- QoS negotiation.
- Mobility management, including roaming.
- Automatic establishment of roaming agreements.

4.7 PLMN Architecture

The network is logically divided into a radio access network and a core network, connected via an open interface. From a functional point of view the core network is divided into a Packet Switched CN Domain, IP Multimedia (IM) CN subsystem [27] and a Circuit Switched CN Domain. IM CN subsystem utilises PS CN domain bearer services.

CS CN domain supports bearer independent transport. There is no difference in service offering or UE functionality due to different transport.

A PS only 3GPP core network is possible as defined within the specification for the Evolved Packet System (EPS) [42].

For further information see 3GPP TS 23.221 [20].

4.8 Interworking Between PLMN and Wireless LANs

Aspects related to interworking between PLMN and WLAN are captured in TS 22.234 [35].

4.9 Network Sharing

Network sharing shall be transparent to the user.

The specifications shall support both the sharing of:

(i) radio access network only;

(ii) radio access network and core network entities connected to radio access network

Note: In a normal deployment scenario only one or the other option will be implemented.

It shall be possible to support different mobility management rules, service capabilities and access rights as a function of the home PLMN of the subscribers.

4.10 The Evolved Packet System

Evolved Packet System is an evolution of the 3G UMTS characterized by higher-data-rate, lower-latency, packet-optimized system that supports multiple RATs. The Evolved Packet System comprises the Evolved Packet Core together with the evolved radio access network (E-UTRA and E-UTRAN).The service requirements for the Evolved Packet System are specified in TS22.278 [42].

4.11 User Data Convergence

4.11.1 Introduction

The UDC concept [45] supports a layered architecture, separating the data from the application logic in the 3GPP system, so that user data is stored in a logically unique repository allowing access from core and service layer entities, named application front-ends. And such unique repository shall be possible to be shared among different PLMNs that have trusted relationships.

Network elements and functionalities should be designed to access profile data remotely and without storing them permanently locally, i.e. the front-ends shall work in a subscriber dataless configuration.

In some cases, services may depend on user data scattered over UDC and other network elements. UDC may support the ability to access necessary network elements to fetch user data on behalf of these services, while minimizing impacts on existing Network Elements in which the data is located.

Applications can subscribe to specific events which will likely occur on specific user data, and those should be notified when those events do appear. The events can be changes on existing user data, addition of user data, and so on.

Third party applications and non trusted network elements should only be able to access the user data after proper authentication and authorization taking into account security and privacy requirements, i.e. it should be possible to present different views on the data to the parties which require access, dependent on the authorization. UDC concept is backwards compatible with 3GPP systems, i.e. it does not have an impact on traffic mechanisms, reference points and protocols of existing network elements.

The UDC concept preserves user authentication and authorization of services across domains, ensuring secured users' access to network.

4.11.2 Management of user data

Due to the logical centralization of user data, it is necessary for UDC to support the provisioning of the user data, that is, user data manipulation like creation, deletion, reading, modification and other operations. Provisioning shall be possible via an external system, self care or dynamically via applications offering e.g. user service configuration facilities.

Operations carried out in the framework of UDC shall support the ACID (Atomicity, Consistency, Isolation, and Durability) characteristics.

4.11.3 User Data Modelling

User Data Modelling refers to the different models that apply to user data convergence: Information Models and Data Models.

An Information Model denotes an abstract, formal representation of entity types, including their properties and relationships, the operations (e.g. read, write…) that can be performed on them, and related rules and constraints. In the information model, entities might have network topology relationship with each other.

In order to accommodate multiple applications and services, existing and new ones, a common baseline information model shall be developed and shall, at minimum, clearly distinguish a number of concepts as entity types:

- Subscriber with relation to several users (e.g. a company and its employees),
- A user attached to different subscriptions (e.g. for a private and a professional service usage)
- A user using multiple devices (e.g. mobiles or fixed)
- Grouping of users to certain categories
- A particular user as member of a certain group
- Service providers' services provided by network operators
- Enterprise services provided by network operators

The baseline information model shall be future proof. It shall not be tied to any specific implementation of the data base or its interfaces. It shall provide flexibility (in its data structure and content), extensibility and multi-application approach.

By extensible, it shall be understood that new applications and/or new service profiles can be added by the operator, if necessary. The flexibility shall permit new data for existing applications to be introduced, or modified.

Data Models are practical implementations of the information model, e.g. Tree-like modelling. The common information model shall allow deriving one or more data models. A reference data model shall be standardized for the message exchange over Ud interface [50], in order to enable multivendor interoperability.

Each application shall only interface the UDC for the data it is dealing with, and not be impacted by other data that UDC stores for other applications. It corresponds to the concept of a data view specific to a given application.

An application can allow access by other applications to data for which it is responsible. This can be done under certain constraint customized by operators.

Access to the UDC data shall be independent of the structure of the data models, i.e. the changes in the data models shall not affect the interface.

5 Evolution

5.1 Support of 2G services

The 3GPP specifications shall be capable of supporting existing 2G services in a manner which is transparent to the users of these services.

5.2 Provision and evolution of services

Since a phased approach has been adopted, the same general service principles shall apply to each phase. Support of services from an end user perspective is understood to be an important driver for established mobile users to stay with their existing operator while taking the new services into use. It is therefore important to enable operators to offer continued support of legacy services in future releases. Previous release services shall as a principle also be supported in the following releases.

Networks shall be capable of providing a specified core set of capabilities.

The core set of capabilities should permit home environment to offer a range of distinctive services including those which cannot be implemented on systems based on previous release specifications.

It shall be possible for the home environment to develop services with full roaming capability.

The radio interface should not unnecessarily restrict the development of new services (within physical limitations).

The standard shall provide a mechanism which allows a terminal to be easily upgraded so that it can access new services which are within the physical limitations of the terminal.

6 Classification of services

In the CS CN domain, the basic services are divided into circuit teleservices (3GPP TS 22.003 [14]) and bearer services (3GPP TS 22.002 [21]) and they can utilise standardised supplementary services (3GPP TS 22.004 [5]).

The PS CN Domain provides IP bearer services. SMS, USSD and UUS can also be considered as bearer services for some applications.

IP multimedia services are the IP based session related services, including voice communications. IP multimedia sessions use IP bearer services provided by the PS CN Domain.

Value added non-call related services include a large variety of different operator specific services/applications. They are usually not specified by 3GPP. The services can be based on fully proprietary protocols or standardised protocols outside 3GPP.

In order to create or modify the above services (both call and non-call related services) operators may utilise toolkits standardised by 3GPP (such as CAMEL or LCS) or external solutions (e.g. Internet mechanisms). Pre-paid is an example of an application created with toolkits that may apply to all of the above services categories.

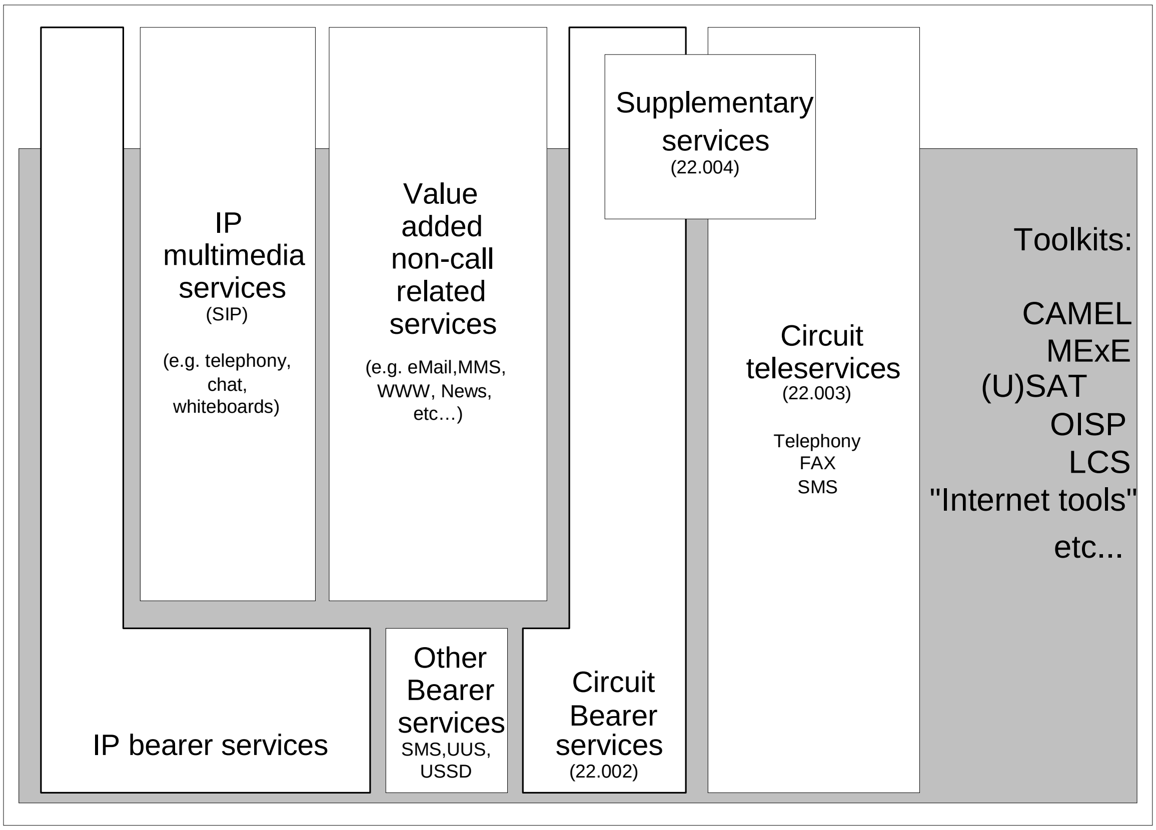

Figure 1: Service classification

7 Principles for new service capabilities

7.1 General

3GPP specifications shall enable the user of a single terminal to establish and maintain several connections simultaneously. It shall efficiently cater for applications which have variable requirements relating to specific QoS parameters (e.g. throughput) whilst meeting other QoS targets. It shall also cater for applications which are able to take adapt to a range of variations in QoS.

7.2 Multimedia

3GPP specifications shall support development of multimedia services and provide the necessary capabilities.

Multimedia services combine two or more media components (e.g. voice, audio, data, video, pictures, text) within one call. A multimedia service may involve several parties and connections (different parties may provide different media components) and therefore flexibility is required in order to add and delete both resources and parties.

Multimedia services are typically classified as interactive or distribution services.

Interactive services are typically subdivided into conversational, messaging and retrieval services:

Conversational services are real time (no store and forward), usually bi-directional where low end to end delays (< 100 ms) and a high degree of synchronisation between media components (implying low delay variation) are required. Video telephony and video conferencing are typical conversational services."

Messaging services offer user to user communication via store and forward units (mailbox or message handling devices). Messaging services might typically provide combined voice and text, audio and high-resolution images.

Retrieval services enable a user to retrieve information stored in one or many information centres. The start at which an information sequence is sent by an information centre to the user is under control of the user. Each information centre accessed may provide a different media component, e.g. high resolution images, audio and general archival information.

Distribution services are typically subdivided into those providing user presentation control and those without user presentation control.

Distribution services without user control are broadcast services where information is supplied by a central source and where the user can access the flow of information without any ability to control the start or order of presentation e.g. television or audio broadcast services.

Distribution services with user control are broadcast services where information is broadcast as a repetitive sequence and the ability to access sequence numbering allocated to frames of information enables the user (or the user's terminal) to control the start and order of presentation of information.

7.2.1 Circuit Switched (CS) multimedia calls

CS multimedia call is a Bearer Service which utilises Synchronous Transparent Data service. The following basic requirements shall be supported for CS multimedia calls [24]:

- CS multimedia call shall be based on a 3GPP specific subset of H.324M.
- All call scenarios shall be supported, i.e. Mobile Originating and Mobile Terminating call against Mobile, ISDN and PSTN call party.
- Single and multiple numbering schemes shall be supported.
- Fallback to speech (TS 11 [14]) shall be supported from 3.1kHz Ext. PLMN multimedia bearer, i.e. if setup of the multimedia call fails the call will be set up as a speech call.
- Service change and fallback shall be supported for UDI/RDI multimedia bearer and speech, to allow fallback to a less preferred service if the preferred service is unsupported, and to change the service between speech and multimedia during the call.

- In the case where a CS multimedia call includes speech (e.g. video call) then the following requirements apply:
  - A user shall be able to change between a speech and CS multimedia call, when desired.
  - When the CS multimedia call is no longer supported, for example due to degraded coverage conditions (including UTRAN to GERAN only transitions), service change shall occur automatically from a CS multimedia call to speech.
  - When a CS multimedia call can be supported, for example due to improved coverage conditions (including GERAN only to UTRAN or UTRAN/GERAN transitions), service change back to the CS multimedia call may be initiated by the network.
  - Other services than CS multimedia call may exist which utilise the Synchronous Transparent Data service. Service transition to/from speech described for CS multimedia call in this clause shall only apply to CS multimedia call and not Synchronous Transparent Data services in general.
- Different bitrates as specified at 3GPP TS 22.002 [21] shall be supported.
- Supplementary services apply to multimedia calls as for Synchronous Transparent Data service according to 3GPP TS 22.004[5].
- When accepting a multimedia call, the user shall be able to request a service change to speech before the call is answered, such that the multimedia path is never actually connected through to the user's phone.
- The user shall be able to deny a service change to multimedia during the call.

7.2.2 IP multimedia (IM) sessions

IP multimedia services are not the evolution of the circuit switched services but represent a new category of services, mobile terminals, services capabilities, and user expectations. Any new multimedia service, which may have a similar name or functionality to a comparable standardised service, does not necessarily have to have the same look and feel from the user's perspective of the standardised service. Voice communications (IP telephony) is one example of real-time service that would be provided as an IP multimedia application.

The following basic requirements are be supported for IP multimedia [27]:

- IP multimedia session control shall be based on SIP [28].
- All session scenarios shall be supported; i.e. Mobile Originating and Mobile Terminating sessions against Internet/Intranet, CS or IM Mobile, ISDN, PSTN call party.
- MSISDN and SIP URL numbering and addressing schemes shall be supported.
- IP multimedia applications shall as a principle, not be standardised, allowing service provider specific variations.

7.2.3 Multimedia Messaging Service (MMS)

The following basic requirements are be supported for MMS:

- Store-and-forward multimedia messaging service with mobile and non-mobile users [25].
- MMS shall be capable of supporting integration of different types of messaging (e.g. fax, SMS, Multimedia, voicemail, e-mail etc.) in a consistent manner.
- Streamed and batch delivery for both message download from the network to the terminal, and messages upload from the terminal to the network.

7.2.4 Real-Time Text Conversation

Real-Time Text (RTT) conversation is a service enabled in 3GPP networks by the Global Text Telephony (GTT) feature [26].

- GTT enables real time, character by character, text conversation to be included in any conversational service, Circuit Switched as well as IP based.
- It is possible to use the text component in a session together with other media components, especially video and voice.
- Interworking with existing text telephony in PSTN as well as emerging forms of standardised text conversation in all networks is within the scope of this feature.
- The text media component can be included initially in the session, or added at any stage during the session.
- The text component is intended for human input and reading, and therefore supports human capabilities in text input speed. The character set support is suitable for the languages the users communicate in.
- GTT specifies limited interoperation with Multimedia Messaging Services including a possibility to divert to messaging in case of call failure and sharing user interface equipment and external UE interfaces.

7.2.5 Packet Switched Streaming Service

The following basic requirements are to be supported for streaming :

- The streaming service uses a client / server model which is transparent to the PLMN. The client controls the initiation and execution of the service.
- The streaming service [30] shall use existing standards (codecs and protocols [31]) where these are available.
- The streaming service utilises the PS Domain with the QoS requirements as specified in 3GPP TS 22.105 [1].

7.3 Service Management Requirements

3GPP specifications shall include standardised protocols enabling service management. It shall enable control, creation and subscription of service capabilities and services, and the management of user profiles.

7.4 Automatic Device Detection

The home environment should be automatically notified when a user, identified by a SIM/USIM, has changed ME and should be informed of the identity of the new ME. This should be applicable to any ME. It should also be possible to achieve Automatic Device Detection for users using any SIM/USIM.

Note: The purpose of this is to enable an automatic configuration of terminals by the operator for specific applications/services if so needed. The procedure for such an automatic configuration need not to be standardized by 3GPP.

The notification that a user has changed ME shall be given as early as possible.

8 Service architecture

In order to provide standardisation of service capabilities a service architecture shown by Figure 2 is envisaged

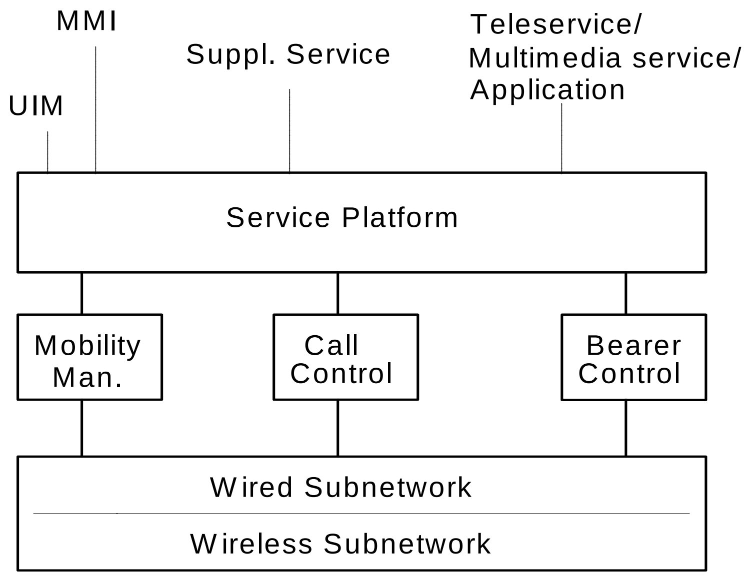

Figure 2: Service Architecture

Figure 2: Service Architecture

A number of bearers shall be provided that can differ in flexibility and offer different capabilities. Bearers may be characterised by parameters such as "throughput", "delay tolerance", "maximum bit error rate", "symmetry" etc. These bearers enable information to be transferred appropriate to the provision of teleservices, multimedia services and end user applications generally, via subnetworks which typically provide different specified qualities of service.

The assignment and release of bearers is provided by the bearer control function. Provision should be made for several bearers to be associated with a call and for bearers to be added to a call and/or to be released from a call following call establishment. The bearers should be independent of radio environments, radio interface technology and fixed wire transmission systems.

Adaptation/Interworking functions are required in order to take account of the differences between the bearers used for the provision of a teleservice/multimedia service/application in the fixed network and the bearers. Adaptation/Interworking functions are required which take account of the discontinuous and/or asymmetrical nature of most teleservices/multimedia services/applications.

The service platform shall provide interfaces (to serving networks and home environments) appropriate to the support, creation and control of supplementary services, teleservices, multimedia services and user applications. The service platform will also provide interfaces enabling subscribers to control supplementary services, teleservices, multimedia services and user applications.

Supplementary service provision and control will be independent of radio operating environment, radio interface technology and fixed wire transmission systems.

As far as possible, the service platform is required to enable new supplementary services, teleservices, multimedia services and/or end user applications to be supported at minimum cost, with minimum disruption of service and within the shortest possible time.

9 Quality of Service (QoS)

The Quality of Service (QoS) parameters should be identified together with appropriate parameter values which set targets to be reached when designing 3GPP specifications, and which also will serve as guidelines for network design and service provision.

The QoS for call set-up time, as an example, can be defined in terms of a mean value and as a percentage of cases which should not exceed a certain time limit. Further information can be found in 3GPP TS 22.105 [1].

The performance requirements for the All-IP Network can be found in 3GPP TS 22.278 [42].

For UE initiated QoS control OMA device management shall be the primary method for provisioning QoS parameters.

10 Emergency Calls

10.1 General requirements

It shall be possible to establish an emergency speech call or GTT [26] call (subject to national requirements). The term 'Emergency call' henceforth refers to speech calls, and GTT Emergency calls if applicable. The term "other media" henceforth refers to media other than speech and GTT. Support of other media types during an emergency call when the IM CN subsystem is used is referred to as 'IMS Multimedia Emergency Session' (MES) and is specified in subclause 10.4.2. Emergency calls will be routed to the emergency services in accordance with national regulations for where the subscriber is located. This may be based upon one or more default emergency call numbers stored in the ME. It shall be allowed to establish an emergency call without the need to dial a dedicated number to avoid the mis-connection in roaming case, such as menu, by use of a 'red button', or a linkage to a car air bag control. Emergency calls shall be supported by the UE without a SIM/USIM/ISIM being present. No other type than emergency calls shall be accepted without a SIM/USIM/ISIM.

Emergency calls shall be supported by UEs that are subject to service restrictions, e.g. for UEs camping on a cell in a forbidden PLMN or in a forbidden LA (see 3GPP TS 22.011 [11]), or on a CSG cell without the subscriber being a member of that CSG (see 3GPP TS 22.220 [48]). Such emergency calls shall be accepted by the network if required by local regulation.

The Emergency service is required only if the UE supports voice.

Note 1: It will be left to the national authorities to decide whether the network accepts emergency calls without the SIM/USIM/ISIM.

It shall be possible to initiate emergency calls to different emergency call centres, depending on the type of emergency. The following types of emergency calls shall be possible:

- Police
- Ambulance
- Fire Brigade
- Marine Guard
- Mountain Rescue
- Manually Initiated eCall (MIeC)
- Automatically Initiated eCall (AIeC)
- Spare

When a SIM/USIM is present, subscriber specific emergency call set-up MMI shall be provided. The Home Environment operator shall specify preferred emergency call numbers (e.g. 999 for UK citizens or 110, 118 and 119 for Japanese citizens). These emergency call numbers shall be stored in the SIM/USIM and the ME shall read this and use any entry of these digits to set up an emergency call. It shall be possible to store more than one instance of this field.

Note 2: Release '98 and earlier SIM cards have the capability to store additional emergency call numbers. However in many cases this has not been used.

It shall be possible to tie any emergency call number to any single emergency call type or to any combination of emergency call types. The association between emergency call numbers and emergency call type shall be able to be programmed by the Home Environment operator into the SIM/USIM.

Example:

19 Police (Albania)

100 Police and Fire Brigade (Greek cities)

100 Ambulance and Fire Brigade (Belgium)

112 Police and Ambulance (Italy)

112 General emergency call, all categories (Sweden)

115 Fire Brigade (Italy)

144 Ambulance (Austria)

If the UE does not recognise the emergency call numbers but the serving network recognises the dialled number as an emergency call number used in the country, a normal call set up shall take place over the radio interface and after the serving network has recognised the emergency number the call shall be routed as an emergency call.

The user friendly MMI that specifies the type of emergency call directly (e.g. menu) should be supported for use in any (i.e. home or visited) PLMN to avoid the mis-connection in roaming case. This shall be allowed both with and without SIM/USIM being present.

When emergency call establishment is initiated, the emergency call type shall be sent by the UE if it is available.

The serving network may download emergency call numbers to the UE in order to ensure that local emergency call numbers are known to the UE. The UE shall regard these emergency numbers as valid in that country only (as identified by the MCC) and shall discard them when a new country is entered.

Note 3: The UE can inform the user if the emergency call type for an emergency number received from the serving network differs from that configured on the USIM/SIM for the same number. How this is implemented is outside the scope of 3GPP and takes into consideration operator policy and regulatory requirements.

If permitted by local regulation, it shall be possible for the user to prevent the sending of his public user identifiers and the location information to the PSAP (i.e. emergency response centre).

Note 4: Operator policies (e.g. requirements for support of emergency communications) may over-ride the user request for suppression.

Emergency sessions over WLAN shall be supported as above with the following caveats:

- The UE issues an Emergency session over WLAN access to EPC when 3GPP access for emergency call is not possible or available (e.g. no 3GPP coverage).
- Only UEs that have a valid subscription, are authenticated, and are authorized are allowed to request an Emergency session over WLAN,
- UEs shall only be able to establish an Emergency session over WLAN when the UE has been provisioned with the WLAN access parameters
- Emergency sessions over WLAN shall only be supported for deployments using Untrusted WLAN procedures,
- Service continuity during Emergency sessions between WLAN and 3GPP access is not supported,
- Requirements for providing the UE location for Emergency sessions over WLAN in a guaranteed and secure fashion are not considered.

10.1.1 Identification of emergency numbers

The ME shall identify an emergency number dialled by the end user as a valid emergency number and initiate emergency call establishment if it occurs under one or more of the following conditions. If it occurs outside of the following conditions, the ME should not initiate emergency call establishment but normal call establishment. Emergency number identification takes place before and takes precedence over any other (e.g. supplementary service related) number analysis.

a) 112 and 911 shall always be available. These numbers shall be stored on the ME.

b) Any emergency call number stored on a SIM/USIM when the SIM/USIM is present.

c) 000, 08, 110, 999, 118 and 119 when a SIM/USIM is not present. These numbers shall be stored on the ME.

d) Additional emergency call numbers that may have been downloaded by the serving network when the SIM/USIM is present.

10.1.2 Domains priority and selection for UE attempts to emergency call

10.1.2.1 Voice and GTT only

A UE that is connected to a domain in which it is possible for the UE to make non emergency calls using the particular media requested by the user, shall use that domain to attempt an emergency call unless serving network policy (based on regulatory requirements and operator needs) requires the UE, including an unauthenticated UE, to attempt the emergency call on a specific domain first.

If the UE is connected to more than one domain in which it is possible for the UE to make non emergency calls using the particular media requested by the user, the UE shall attempt an emergency call on the same domain it would use to originate a non-emergency call using the same media unless serving network policy (based on regulatory requirements and operator needs) requires the UE, including an unauthenticated UE, to attempt the emergency call on a specific domain first.

In the case where an emergency call attempt by a UE fails, the UE should automatically make a second attempt on the other domain if the UE supports it.

If the user aborts the emergency call setup during the subsequent automatic attempt and immediately tries to set up an emergency call again, then the UE shall immediately attempt in the domain in which the user aborted the emergency call.

10.1.2.2 Other media

The following applies in addition to 10.1.2.1.

If an emergency call attempt that includes a request for both (i) voice and/or GTT and (ii) other media cannot be supported or fails in all connected access types in the PS CN domain, the UE shall attempt the emergency call in the CS domain if available and shall only include the request for voice and/or GTT.

10.1.3 Call-Back Requirements

Subject to local/regional regulations the network shall support a call-back from a PSAP.

It shall be possible to supply the user's Directory Number/MSISDN/SIP URI as the CLI to the PSAP to facilitate call-back. The CLI used on call-back shall allow the PSAP to contact the same terminal that originated the emergency call.

If the incoming call can be identified by the core network as a call-back to an emergency call (i.e. coming from a PSAP) then supplementary services at the terminating party shall be handled as described in TS 22.173 [40] for Multimedia Telephony (e.g. Communication Diversion, Communication Hold, Communication Barring).

Note: There is no specific callback requirement for CS supplementary services.

A call-back may be attempted for a period of time defined by local regulations after the emergency call release. In case of a UE in limited service state, call-back is not required.

10.2 Emergency calls in the CS CN Domain

A CS CN Domain shall support the emergency call teleservice as defined in 3GPP TS 22.003 [14] (TS12).

If a UE supports TS11(Telephony)[14], then it shall also support TS12(Emergency Calls)[14]. It shall be possible to set up emergency calls initiated by an emergency call number.

10.3 Emergency Calls in the PS CN Domain

Without the IM CN subsystem, emergency calls are not supported in the PS CN domain.

10.4 Emergency calls in the IM CN subsystem

10.4.1 General

The IM CN subsystem shall support IMS emergency calls. It shall be possible to set up emergency calls initiated by an emergency call number.

If a UE supports IMS Multimedia Telephony service with speech media as specified in TS 22.173 [40] via an access network, then it shall also support IMS emergency calls via that access network.

Subject to the regulatory requirement, the IM CN subsystem shall be able to unambiguously identify each emergency service defined in the national numbering plan for the country in which the UE is located.

In accordance with national regulations for where the subscriber is located, if the UE does not recognise a dialled number as an emergency call number but the IM CN where the subscriber is located does recognise the dialled number as an emergency call number (e.g. a number used in the local emergency numbering plan) then the call shall be routed as an emergency call indicating the type of emergency service to the correct PSAP. Subject to operator setting the call may be prioritized.

Note 1: The above does not preclude the network rejecting the call and requesting the UE to setup a new emergency call to the same emergency service.

Emergency calls may be initiated using a private numbering plan [49].

Note 2: There can be an overlap between the private numbering plan of a hosted enterprise and the public numbering plan, which makes translation of emergency numbers necessary.

Emergency calls may be initiated by a service when requested by the user.

Note 3: It is not intended to enable automatic setup of emergency calls.

Note 4: Only speech and GTT-IP [47] media are supported, when required per clause 10.1, for emergency services towards a CS PSAP.

An emergency call shall take precedence over any other services a UE may be engaged in, if required by local regulation.

Emergency calls from an unauthenticated UE (as far as the IM CN is concerned) shall be supported by the IM CN subsystem, if required by local regulation.

Subject to regulatory requirements, when UEs must be authenticated, both the network and the UE shall support the same authentication and security methods that are used for non-emergency sessions.

10.4.2 IMS Multimedia Emergency Sessions

10.4.2.1 General

For IMS emergency calls towards IP PSAPs, other media types may be supported by the UE and the IMS, subject to regulatory requirements.

The media types that may be supported during an IMS MES include:

- Real time video (simplex, full duplex), synchronized with speech if present;
- Session mode text-based instant messaging;
- File transfer;
- Video clip sharing, picture sharing, audio clip sharing;
- Voice; and
- Real-Time Text.

Note 1: An IMS MES need not contain voice or Real-Time Text.

To avoid interworking issues, a UE and IMS that supports text based instant messaging shall support a common session mode text-based instant messaging protocol.

IMS MES does not include support for legacy store and forward messaging such as the Short Messaging Service (SMS).

Calls from non-human associated devices (e.g. fire alarms) are outside the scope of this specification.

Adding, removing and modifying individual media to/from an IMS MES shall be supported.

An IMS MES is not a subscription service. A UE capable of IMS emergency calls and capable of supporting the other media types should also be able to support initiation of an IMSMES.

An IMS MES from an unauthenticated UE (as far as the IM CN is concerned) shall be supported by the IM CN subsystem, if required by local regulation.

IMS MES shall be supported by UEs that are subject to service restrictions, e.g. for UEs camping on a cell in a forbidden PLMN or in a forbidden LA (see 3GPP TS 22.011 [11]), or on a CSG cell without the subscriber being a member of that CSG (see 3GPP TS 22.220 [48]). Such IMS MES shall be accepted by the network if required by local regulation.

An IMS MES shall support providing the location of the UE, in a manner similar to IMS emergency voice calls.

An IMS UE that supports IMS MES shall identify an emergency number dialled by the end user as a valid emergency number utilizing the same mechanisms as used for IMS emergency voice calls as defined in subclause 10.1.1.

Note 2: This capability supports the general public, including facilitating emergency communications by individuals with disabilities (e.g. persons who are deaf, deaf-blind, hard of hearing, or have a speech disability).

An originating network and UE may support some or all of these other media types, and support of any specific media by an originating network or UE may be subject to regulatory requirements.

Voice call continuity per clause 21 shall be supported when a UE with an active IMS MES with voice and other media moves out of IMS voice coverage and voice call continuity is supported by the UE and network. The remaining media (i.e. voice call) then becomes a CS emergency call e.g. TS12 call for 3GPP systems as defined in 3GPP TS 22.003 [14].

Other media shall be dropped when a UE with an active IMS MES moves out of IMS voice coverage, irrespective of whether or not there is an active voice session.

10.4.2.2 UE Requirements

When IMS MES are supported by the UE, the following apply:

- An IMS UE that supports IMS MES shall also support IMS emergency voice calls.
- Once a UE is aware that an IMS MES has been initiated, the UE shall be able to (subject to user configuration) avoid drawing unnecessary attention to the user (e.g., playing audible tones or flashing brightly) and should confirm this to the user in as private a manner as is reasonable e.g. using text on the screen or audio if headphones are already connected. UE behaviour in an IMS MES may need to be different relative to the normal configuration.
- The UE should clearly differentiate IMS emergency session-mode text based instant messaging from IMS non-emergency session-mode text based instant messaging on the user display.
- The IMS UE supporting video transfer during an IMS MES should be able to deliver recorded media in a form that allows progressive playback. (It is desirable that all pre-recorded media sent during an emergency session be progressively viewable.)
- When an IP PSAP attempts to add additional media to an existing IMS MES, the user shall be made aware of this. When additional media is requested by the PSAP, the user shall be able to permit or deny it.

- The UE shall provide an indication to the user for each requested media, whether it was successfully or unsuccessfully established.
- Further notifications of added and removed media shall be indicated to the user while the IMS MES is active.
- If none of the media requested by the UE is successfully established, the IMS MES will fail and an IMS MES failure indication shall be provided to the user.
- In handover of an IMS MES where other media is dropped when IMS MES is not supported, the UE shall indicate to the end user that the other media is not supported in this area.

The following requirements for IMS emergency voice calls also apply when an IMS MES is supported by the UE:

- An IMS UE that supports IMS MES shall indicate to the network that the call is an IMS emergency call as is done for an IMS emergency voice call.
- An IMS UE that supports IMS MES shall be able to receive an IMS call-back from a PSAP per clause 10.1.3 with voice, GTT or other media per clause 10.4.2.1.
- An IMS UE that supports MES shall utilize the same trust and security mechanisms for the other media as utilized for an IMS emergency voice call.
- When roaming, a UE shall originate an IMS MES in the serving network in the same manner as for IMS emergency voice calls.

10.4.2.3 Originating Network Requirements

When an IMS MES is enabled by the originating network, the following apply:

- Other media shall only be supported in packet-based networks that support IMS emergency voice calls.
- The originating network shall deliver all media to the same IP PSAP throughout the duration of the IMS MES.
- The network shall indicate to the UE, for each requested media, whether it was successfully or unsuccessfully established.
- Further notification of added and removed media shall be provided to the UE while the IMS MES is active.
- If none of the media requested by the UE is successfully established, the IMS MES will fail and an IMS MES failure indication will be provided to the UE.

The following requirements for IMS emergency voice calls also apply when IMS MES is supported by the network:

- Subject to regional regulatory requirements, the network shall be able to authenticate the UE using the same procedures as for IMS emergency voice calls.
- The originating network shall provide the capability to enable an IMS UE supporting IMS MES to obtain local emergency numbers or other emergency address(es) (e.g. destination address) utilizing the same mechanism as used for IMS emergency voice calls.
- An IMS MES shall be provided in the local serving network.
- For an IMS MES, any kind of emergency addressing (e.g. SIP URIs, Tel URIs) and special indications for emergency sessions shall be treated in the same manner as IMS emergency voice calls.
- The originating network should detect all IMS MESs regardless of the UE emergency call indication. According to operator policy, the originating network may either inform the UE to enable re-origination as an IMS MES or support origination of the initial call.
- The originating network shall be responsible for routing the IMS MES towards the appropriate PSAP (e.g., based on emergency service type, location, or policy).
- The network shall be able to provide integrity protection, and/or privacy for other media similar to that provided for IMS emergency voice calls.
- An IMS MES shall utilize the same priority mechanisms as IMS emergency voice calls.

- Detailed log records of the IMS MES shall be generated by the originating network in a similar manner to IMS emergency voice calls and subject to regulatory requirements.
- All media content within the IMS MES shall be carried with an indication of the source, in a similar manner as for IMS emergency voice calls.

10.5 Void

10.6 Location Availability for Emergency Calls

National regulations may require wireless networks to provide the emergency caller's location. This requirement typically overrides the caller's right to privacy with respect to their location being revealed, but remains in effect only as long as the authorities need to determine the caller's location. The interpretation of the duration of this privacy override may also be different, subject to national regulation. For example, some countries require location to be available from the wireless network only while the call is up, while others may allow PSAP's to unilaterally decide how long the location must be made available.

Therefore, the requirement for providing location availability is to allow the network to support providing a mobile caller's location to the authorities for as long as required by the national regulation in force for that network.

Note: See TS 22.071 [44] for location service requirements on emergency calls.

10.7 Transfer of data during emergency calls

Emergency calls may be supplemented with emergency related data [1]. Typically this data enables the accurate geographic location of a manually or automatically activated emergency calling device e.g. an in vehicle system (IVS), to be provided to the Public Safety Answering Point (PSAP).

The following requirements apply to UEs designed to be able to perform transfer of data during an emergency call and to networks supporting transfer of data during an emergency call:

- The data may be sent prior to, in parallel with, or at the start of the voice component of an emergency call.
- Should the PSAP request additional data then this may be possible during the established emergency call.
- The realisation of the transfer of data during an emergency call shall minimise changes to the originating and transit networks.
- Both the voice and data components of the emergency call shall be routed to the same PSAP or designated emergency call centre.
- The transmission of the data shall be acknowledged and if necessary data shall be retransmitted.
- The UE shall indicate at call setup if the emergency call will carry supplementary data.

UEs designed to be able to perform transfer data during emergency calls and configured to only perform emergency calls with transfer of data (eCall only mode) shall comply with the following additional requirements:

- The UE shall not perform mobility management procedures, including registration on a PLMN, except when attempting to initiate and during an emergency call, or to initiate a test or reconfiguration of the terminal upon request from the user..
- For UEs that have the ability to be called back by the PSAP, the UE shall be capable to continue mobility management procedures for a limited duration following the termination of the eCall.
- The UE shall contain an USIM application.
- In the case where the user subscribes to other services provided by the PLMN, it shall be possible for the network operator to reconfigure the UE so that it can access the subscribed services.
- It shall be possible for the user of the UE to change network operator/service provider (i.e. to use a different USIM) or for the subscriber to modify the existing subscription used with the UE.

- It shall be possible for the UE upon request from the user to initiate a call to an operator designated non-emergency MSISDN for the purpose of accessing test and terminal reconfiguration services.

Additional national and regional requirements are as specified in Annex A.

10.8 Supplementary service interaction during emergency calls

Supplementary services that interrupt or divert the media path between a PSAP and the end device shall be handled as specified in TS 22.173 [40] (e.g. Communication Hold) for Multimedia Telephony. No such Supplementary Services are applicable to CS Emergency Calls (TS12) according to TS 22.004 [5].

11 Numbering principles

The following network addressing schemes listed below shall be supported at the relevant domains:

- E.164,
- E.168,
- E.212,
- X.121
- Internet (including e.g. IP address, SIP URI).

11.1 Number portability

11.1.1 Requirements for CS CN domain

Some numbering schemes shall be fully independent of the supporting serving network and the home environment, allowing users to transfer this number to another home environment. For further information see 3GPP TS 22.066 [7].

An MSISDN shall be allocated to each new user at the start of a subscription. This number may be allocated from one of several numbering domains. For example:

- home / serving environment numbering scheme;
- national numbering scheme;
- regional numbering scheme;
- global numbering scheme.

A user shall be able to move subscription from one home environment to another without changing the MSISDN provided that the new home environment offers service in the same geographic domain. It is envisaged that home environment s will be able to allocate MSISDNs from each of these domains as required.

11.1.2 Requirements for PS CN domain

None identified.

11.1.3 Requirements for IM CN subsystem

It shall be possible to offer number portability for E.164 numbers within IM CN subsystem. For further information see 3GPP TS 22.066 [7].

11.2 Evolution path

Since 3GPP specifications aim to be aligned with IMT-2000, a primary goal in numbering is the provision of global user numbering in line with steps taken by the ITU - SG2.

The numbering scheme and network implementation chosen shall allow for international/global evolution.

11.3 Void

11.4 Void

11.5 Void

11.6 Private numbering

A user may wish to use private numbers for the purposes of calling frequent numbers. Therefore there is a requirement for the use, by the user, of Private Numbering Plans (PNPs). These schemes may belong to the user himself, to a home environment or a third party.

11.7 Numbering schemes

11.7.1 Multiple numbering scheme

The standards shall support the possibility of allowing the bearer service associated with an MT call to be implicitly defined by the destination MSISDN, for example to use a different MSISDN to establish voice, fax or data . It will be possible for multiple MSISDNs to be associated with a single subscription.

11.7.2 Single numbering scheme

The standards shall support the possibility of allowing MT calls of different bearer types (e.g. voice, fax, data) to be routed to a single MSISDN. It is recognised that the implementation of this may depend on the availability of bearer information associated with an incoming call from the adjoining transit network. In particular the standards will support this possibility in the case of an adjoining ISDN transit network.

11.7.3 Additional numbers

The 3GPP system shall support the possibility to assign an additional MSISDN, in addition to the original MSISDN, to a user with a connection to the PS CN domain. If this additional MSISDN is available it shall be used for correlation of CS and IMS in voice call and service continuity as well as IMS Centralized Service. In this case the original MSISDN may be used for charging and OA&M purposes and forwarded to the PS gateway to other packet data networks.

11.8 Optimal routing

The implementation of the numbering scheme used shall allow for optimal routing; i.e. routing shall not take place simply on the number dialled.

See 3GPP TS 22.079 [8] for some scenarios for the CS CN domain. Optimal routing for IP services is supported by the All-IP Network [42].

11a Identification Requirements

11a.1 Subscriber Identification

In 3GPP the identity of a subscriber is encoded in a identity module application which is contained on a UICC or on a GSM SIM card. The UICC or GSM SIM card is a removable component of the User Equipment. Three types of identity modules are used in the 3GPP system:

- Universal Subscriber Identity Module (USIM)
- IMS Subscriber Identity Module (ISIM)
- Subscriber Identity Module (SIM) according to GSM

General requirements:

- In the 3GPP system each subscriber shall be uniquely identifiable.
- The serving networks shall be able to authenticate any subscriber that roams onto their network
- If a UE, that is registered on the serving network, contains a GSM SIM card or a UICC containing a identity module application, the serving network shall be able to identify the associated home PLMN.

Note 1: UE support of GSM SIM is optional.

Note 2: See the chapter (USIM, UICC and Terminal) of the present specification for a reference, which GSM phase SIMs need to be supported by the network.

11a.2 Terminal Identification

It is a requirement that the terminal can be uniquely identified by the home environment and serving network. This shall require a terminal identity scheme which uniquely identifies each terminal, see 3GPP TS 22.016[12].

11a.3 Home Environment / Serving Network Identification

Home / serving environments need to route communication to the current location of the user. This shall require a identity scheme which uniquely identifies the serving environment and shall be used for routing purposes.

11a.4 Serving Environment / Mobile Virtual Network Identification

A mobile virtual network operator (MVNO) is a service provider that does not have its own radio access network, but resells wireless services, typically under their own brand name, using the network of a host PLMN operator.

It should be possible to uniquely identify subscribers belonging to a particular MVNO.

12 Human Factors and user procedures

The User Interface (MMI) from the end-user's point of view should be as flexible as possible while still meeting the general service requirements. In addition it should be capable of being updated so as to meet new services which are still to be envisaged.

In general the following principles should be encompassed:

- activation of services should be as simple as possible with minimum input expected from the user;
- feedback, to the user from the various services, should be meaningful;
- any error recovery procedures provided should be simple to understand and execute.
- input from the user and information to the user should be provided in alternative selectable modes in order to match user capabilities, preferences and situation.

However, a detailed specification for the User Interface shall not be defined. In particular given the global nature of the third generation systems, for different regions of the world, different criteria will determine the implementation of the User Interface. Also it is unlikely that there will be a single common handset which will meet all the service requirements and therefore a common User Interface would be impractical.

Given the flexibility of the services, there should be a wide range of User Interface possibilities. These possibilities include simple terminals with a single on/off button through to complex terminals providing support to hearing/visually impaired users.

Control of CS CN Domain supplementary services (3GPP TS 22.004 [5]), may use MMI procedures specified in 3GPP TS 22.030 [6] and existing MMI related UE features (Annex A) may also be used. In particular the following features are highly desirable for uniform UE implementation where appropriate:

- Mapping of numeric keys to European alphabetic keys to ensure compatible mnemonic dialing as defined in 3GPP TS 22.030 [6],
- "+" key function to enable one key international access as defined in Annex A
- Structure of the MMI as described in 3GPP TS 22.030 [6]
- Presentation of IMEI (International Mobile Equipment Identity) as defined in 3GPP TS 22.030 [6].

13 UICC, USIM and Terminal

This clause defines the functional characteristics and requirements of the User Service Identity Module (USIM) and ISIM (IM Services Identity Module). The USIM/ISIM are applications residing on a UICC.

13.1 The USIM/ISIM and User Profiles

13.1.1 The USIM

Every USIM shall have a unique identity and shall be associated with one and only one home environment.

It shall be possible for a home environment to uniquely identify a user by the USIM.

The USIM shall be used to provide security features.

For access to services, provided by PS or CS CN domains, a valid USIM shall be required. Optionally, SIM according to GSM phase 2, GSM phase 2+, 3GPP release 99, 3GPP release 4 specifications may be supported.

The USIM shall be able to support SIM Application Toolkit as specified in 3GPP TS 22.038 [3].

The USIM shall reside on a UICC. USIM specific information shall be protected against unauthorised access or alteration.

It shall be possible to update USIM specific information via the air interface, in a secure manner.

Access to the IMS services shall be possible using the USIM application in the event of no ISIM being present on the UICC. If an ISIM is present on the UICC it shall be used to access the IMS.

It shall be possible to store provisioning parameters on the USIM according to DM specifications [37].

It should be possible to store provisioning parameters on the USIM according to CP specifications [38].

It shall be possible for the network operator to configure the USIM to indicate (through personalisation and OTA) whether provisioning parameters according to DM specifications or provisioning parameters according to CP specifications shall be used.

Note: To avoid misoperation of the UE in a mixed provisioning environment e.g. during a transition phase when both CP and DM clients are present in the UE, the CP parameters on the USIM can be read first. If DMinformation is present (provisioned OTA in the CP parameters.), then use DM, otherwise use CP.

Annex A describes a number of features that may optionally be supported by the UE and thus USIM.

13.1.2 User Profiles

It shall be possible for a user to be associated with one or a number of user profiles, which the user can select and activate on a per call basis. The user profile contains information which may be used to personalise services for the user.

It shall be possible for one or more user profiles associated with the same user to be active simultaneously so that the user may make or receive calls associated with different profiles simultaneously. Activation of profiles shall be done in a secure manner, for example with the use of a PIN.

For terminating calls the correct profile shall be indicated by the user address used (e.g. MSISDN, SIP URI), each profile will have at least one unique user address associated with it. For originating calls the user shall be able to choose from the available profiles, the appropriate one for the call. A profile identity will need to be associated with the call for accounting and billing purposes. User profile identities need not be standardised but a standardised means is required for indicating that a particular profile is being used.

Simultaneous use of the same user profile on multiple terminals for the same type of service shall not be allowed.

User profiles associated with different home environments shall not share the same user address.

13.1.3 UICC usage in GERAN only Terminals

In Release 5 and later, terminals supporting only GERAN shall support USIM.

Note: It is strongly recommended that manufacturers implement SIM support on GERAN only terminals until the population of SIMs in the market is reduced to a low level.

13.1.4 Multiple USIMs per UICC

The standard shall support more than one USIM per UICC even when those USIMs are associated with different home environments. Only one of the USIMs or the SIM shall be active at a given time. While the UE is in idle mode, it shall be possible for the user to select/reselect one USIM application amongst those available on the UICC. At switch on, the Last Active USIM shall be automatically selected. The Last Active USIM shall be stored on the UICC. By default if there is no Last Active USIM defined in the UICC, the user shall be able to select the active USIM amongst those available on the UICC.

The standard must not prevent the coexistence of USIM applications, each associated with different home environments on the same UICC, so long as the security problems which arise from such a coexistence are solved.

13.1.5 The ISIM

Access to the IMS services shall be possible using an ISIM application.

The ISIM shall be sufficient for providing the necessary security features for the IMS and IMS only.

The ISIM shall reside on a UICC. ISIM specific information shall be protected against unauthorised access or alteration.

It shall be possible to update ISIM specific information via the air interface, in a secure manner.

Note: When accessing IMS over GERAN/UTRAN/EUTRAN or I-WLAN using ISIM, a USIM needs also be present to access the rest of the 3GPP system. Alternatively USIM could be used to access IMS.

13.2 The UICC

Characteristics including physical formats of a UICC are defined in TS 31.101[36].

Access to services via 3GPP system with a single UICC shall be possible.

13.2.1 The UICC and Applications other than the USIM or ISIM

It shall be possible for the UICC to host other applications in addition to the USIM or ISIM, see figure 3. Service providers, subscribers or users may need to establish additional data or processes on the UICC. Each application on an UICC shall reside in its own domain (physical or logical). It shall be possible to manage each application on the card separately. The security and operation of an application in any domain shall not be compromised by an application running in a different domain. Applications may need to use their own security mechanisms which are separate to those specified by 3GPP e.g. electronic commerce applications.

Examples of UICC applications are: USIM, ISIM, off-line user applications like UPT, electronic banking, credit service, etc.

Applications should be able to share some information such as a common address book.

It shall be possible to address applications, which reside on the UICC, via the air interface.

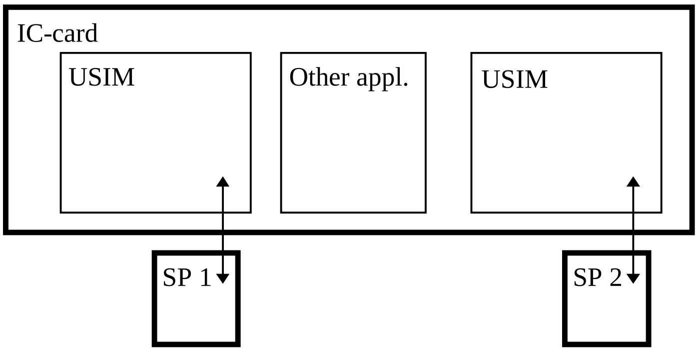

Figure 3 Example of a Multifunction UICC

13.2.1a UICC applications and IMS

UICC applications may make use of IMS functionalities controlled by ME.

Note: This is to allow a UICC application to interact with an Application Server (AS) through IMS. Examples of UICC applications include identity management, banking applications, etc.

13.2.2 Fast Access and Retrieval of Data from UICC

An optional "high speed" interface may be provided between the UICC and the ME.

If provided, this interface shall allow fast access and retrieval of data to support functionalities requiring large amounts of data to be transferred to and from the UICC. Examples include:

- on-card web servers
- rapid access to data stored on the UICC, e.g. phone book, PLMN lists or user data

A UICC/ME interface supporting this "high speed" interface shall be backward compatible with the TS 102 221 interface specified in 3GPP TS 31.101 [36].

13.3 Terminals and Multiple UICCs

A single terminal may support the use of multiple UICC (e.g. with applications like USIM and/or banking, credit card,...). Only one UICC shall be active at a time to access a PLMN. In case the active UICC contains more than one USIM, the requirements of 13.1.4 shall apply.

If the UICC with the active USIM is removed from the mobile terminal during a call (except for emergency calls), the call shall be terminated immediately. If the UICC with an active ISIM is removed during an IMS session the IMS session shall be terminated.

14 Types of features of UEs

3GPP specifications should support a wide variety of user equipment, i.e. setting any limitations on terminals should be avoided as much as possible. For example user equipment like hand-portable phones, personal digital assistants and laptop computers can clearly be seen as likely terminals.

In order not to limit the possible types of user equipment they are not standardised. The UE types could be categorised by their service capabilities rather than by their physical characteristics. Typical examples are speech only UE, narrowband data UE, wideband data UE, data and speech UE, etc..

In order to enhance functionality split and modularity inside the user equipment the interfaces of UE should be identified. Interfaces like UICC-interface, PCMCIA-interface and other PC-interfaces, including software interfaces, should be covered by references to the applicable interface standards.

UEs have to be capable of supporting a wide variety of teleservices, multimedia services and applications provided in PLMN environment. Limitations may exist on UEs capability to support all possible teleservices, multimedia services and information types (speech, narrowband data, wideband data, video, etc.) and therefore functionality to indicate capabilities of a UE shall be specified.

The basic mandatory UE requirements are:

- Support for USIM. Optional support of GSM phase 2, 2+, 3GPP Release 99 and Release 4 SIM cards [34]. Phase 1, 5V SIM cards shall not be supported. Support for the SIM/ISIM is optional for the UE, however, if it is supported, the mandatory requirements for SIM/ISIM shall be supported in the UE;

Note 1: There is no Release 5 specification for the SIM, and therefore references to "SIM" apply to earlier releases.

Note 2: It is strongly recommended that manufacturers implement SIM support on terminals supporting GERAN until the population of SIMs in the market is reduced to a low level.

- Home environment and serving network registration and deregistration;
- Location update;
- Originating or receiving a connection oriented or a connectionless service;
- An unalterable equipment identification; IMEI, see 3GPP TS 22.016 [12];
- Basic identification of the terminal capabilities related to services such as; the support for software downloading, application execution environment/interface, MExE terminal class, supported bearer services.
- Terminals capable for emergency calls shall support emergency call without a SIM/USIM/ISIM.
- Support for the execution of algorithms required for encryption, for CS and PS services. Support for non encrypted mode is required;
- Support for the method of handling automatic calling repeat attempt restrictions as specified in 3GPP TS 22.001 [4];
- At least one capability type shall be standardised for mobile terminals supporting the GERAN,UTRAN and E-UTRAN radio interfaces.
- Under emergency situations, it may be desirable for the operator to prevent UE users from making access attempts (including emergency call attempts) or responding to pages in specified areas of a network, see 3GPP TS 22.011 [11];
- Ciphering Indicator for terminals with a suitable display;
- The ciphering indicator feature allows the UE to detect that the 3GPP radio interface ciphering (user plane) is not switched on and to indicate this to the user. The ciphering indicator feature may be disabled by the home network operator setting data in the SIM/USIM. The default terminal behaviour shall be to take into account the operator setting data in the SIM/USIM. However, terminals with a user interface that can allow it, shall offer the possibility for the user to configure the terminal to ignore the operator setting data in the SIM/USIM. If this feature is not disabled by the SIM/USIM or if the terminal has been configured to ignore the operator setting data

in the SIM/USIM, then whenever a user plane connection is in place, which is, or becomes un-enciphered, an indication shall be given to the user. In addition, if this feature is not disabled by the SIM/USIM or if the terminal has been configured to ignore the operator setting data in the SIM/USIM, then additional information may also be provided about the status of the ciphering. Ciphering itself is unaffected by this feature, and the user can choose how to proceed;

- Support for PLMN selection.

- Support for handling of interactions between toolkits concerning the access to UE MMI input/output capabilities;

- Whenever an application (e.g. a SAT/MExE/WAP application) requires the access to the UE MMI input/output capabilities (e.g. display, keyboard,…), the UE shall grant this access subject to the capabilities of the UE. This shall not cause the termination of any other applications (e.g. WAP browser or MExE/SAT application) which were previously using these UE resources. The UE shall give the user the ability to accept or reject the new application. In the case that the application request is rejected, the access to the UE MMI input/output capabilities is returned to the applications which were previously using these UE resources. If the user decides to continue with the new application, then when this new application is terminated, the access to the UE MMI input/output capabilities shall be returned to the UE to be re-allocated to applications (e.g. the preceding application which was interrupted). Subject to the capabilities of the UE, the user shall have the ability to switch the MMI input/output capabilities between applications.

Note: Rejecting a request to access the UE MMI input/output capabilities by an application does not necessarily mean that it is terminated, but only that the access to the UE MMI input/output capabilities are not granted to this application. Handling of rejection (termination, put on hold,…) is the responsibility of the application.

Annex A describes a number of features which may optionally be supported by the UE.

15 Relationship between subscription and service delivery

15.1 Subscription

A subscription describes the commercial relationship between the subscriber and the service provider.

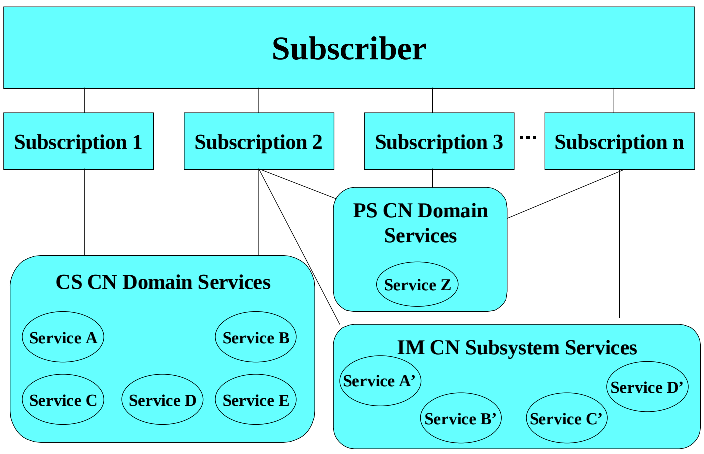

Figure 4: Subscriber, subscription and services relationship

A subscription to a network operator may provide the user with access to one or more domains. A Subscription shall identify the set of services, within particular domains, to which the user has access (see figure 3); each subscription may specify a different set of services. These services may be provided by the CS CN Domain and/or a PS CN Domain and/or an IM CN subsystem. Subscriptions relate to services such as Basic Services (e.g. Teleservices, Bearer services), PS services and IM-Services (IP-based multimedia services), which are typically provided by network operators, and to value added services which typically are provided by network operators and/or other entities that provide services to a subscriber

The subscription identifies:

- the services and related services information that are made available to the subscriber by the service provider ;

In addition a subscription to a network operator may identify:

- the domains to which the user has been granted access by the network operator. In particular, the PS service profile and information on the allowed QoS parameter ranges shall be contained in the subscription.

- the identity of the subscriber within these domains.
  Note: The identity of a subscriber in the CS CN domain and PS CN domain (e.g. her IMSI) may potentially be different to her identity in the IM CN subsystem

- the radio access systems over which the subscriber may access their services e.g. UTRAN, GERAN, EUTRAN, I-WLAN.

15.2 Other concepts associated with services

Provision of services:
An action to make a service available to a subscriber. The provision may be:

- general: where the service is made available to all subscribers (subject to compatibility restrictions enforced) without prior arrangements being made with the service provider;

- pre-arranged: where the service is made available to an individual subscriber only after the necessary arrangements have been made with the service provider.

Withdrawal:
An action taken by the service provider to remove an available service from a subscriber's access. The withdrawal may be:

- general: where the service is removed from all subscribers provided with the service;

- specific: where the service is removed on an individual basis from subscribers provided with the service.

Note: Access to the IM subsystem requires IP connectivity provided, for example, through provision of the PS CN domain.

15.3 Requirements concerning service delivery

In general it is a requirement to allow the use of independent services simultaneously (i.e. Basic, PS, IP multimedia and operator specific).

1) The network usage shall be based on the services identified within the subscription, the terminal capabilities and, where applicable, roaming agreements between operators.

2) The Home environment shall be able to decide on the service delivery in a roaming scenario. I.e. it shall control how services are delivered in line with the subscription.

3) If an offered or required service (e.g. voice) could be provided with different technologies within the serving network, the decision on service delivery shall be based on preferences identified in the user profile and serving network capabilities and conditions (e.g. load).

4) If the user profile does not allow an alternative service delivery method and the requested delivery method is not available in the serving network the service shall not be provided to the subscriber. This applies also to data bearer services with defined QoS parameters (or parameter ranges).

Examples:

- A terminating voice call for a subscriber with a dual/multi mode terminal (e.g. UTRAN/GERAN) could be delivered in a hybrid network as IM service or CS voice call (TS11). The delivery decision is based on the preferences of service delivery within the user profile and the network conditions. If there is no preference information of the Home environment available the decision is made only on the network conditions from the serving network.
- A terminating data service (e.g. PS with QoS for real time audio) where the network cannot provide the QoS at call setup. Both the originating and terminating application shall be informed about the possible QoS configuration for that call. The further handling (setup continuation, termination) depends on the decisions of the applications.

15.3.1 Mobile Originated Voice calls

When a ME capable of offering voice service both on CS and IMS is CS attached and IMS registered MO CS Voice calls (TS11) and MO IMS voice services shall be originated on the domain specified by the Home operator policy or users preferences. The Home Operator policy shall have precedence to user preferences.

15.3.2 Mobile Terminated Voice calls

When a ME capable of offering voice service both on CS and IMS, MT CS Voice calls (TS11) and MT IMS voice services shall be delivered over the domain specified by the Home operator policy or users preferences. The Home Operator policy shall have precedence to user preferences. If the call delivery attempt fails in one domain, if specified by operator policy, it should be possible to attempt the delivery in the other domain or the call/communication forwarding supplementary services [41, 40] may be invoked if provisioned.

Note: The delivery decision will take into account aspects such as IMS registration and CS attachment status.

16 Charging principles

The cost of the call may cover the cost of sending, transporting, delivery and storage. The cost of call related signalling may also be included. Provision shall be made for charging based on time, destination, location, volume, bandwidth, access technology and quality. Charges may also be levied as a result of the use of value added services.

It shall be possible for information relating to chargeable events to be made available to the home environment at short notice. The requirements shall include:

- Immediately after a chargeable event is completed;
- At regular intervals of time, volume or charge during a chargeable event.
- Delivery of the location of the terminal to the home environment, e.g. cell identification;

Standardised mechanisms of transferring charging information are required to make these requirements possible.

It should be possible for multiple leg calls (e.g. forwarded, conference or roamed) to be charged to each party as if each leg was separately initiated. However, in certain types of call, the originating party may wish/be obliged to pay for other legs (e.g. SMS MO may also pay for the MT leg.).

It shall be possible to charge according to the location (e.g. cell, or zone) and access technology that are being used to access network services.

Provision shall be made for the chargeable party to be changed during the life of the call. There shall be a flexible billing mechanism which may include the use of stored value cards, credit cards or similar devices.

The chargeable party (normally the calling party) shall be provided with an indication of the charges to be levied (e.g. via the called number automatically or the Advice of Charge supplementary service) for the duration of the call (even though the user may change service environment)The user shall be able to make decisions about the acceptable level of accumulated charge dynamically or through their service profile.

If a user is to be charged for accepting a call then their consent should be obtained. This may be done dynamically or through their service profile.

Charging and accounting solutions shall support the shared network architecture so that end users can be appropriately charged for their usage of the shared network, and network sharing partners can be allocated their share of the costs of the shared network resources.

17 Roaming

17.1 Assumptions

In order to roam, the following applies:

- Mobile terminal can connect to the radio access network.
- Authentication (charging/billing network) must occur in order to get access to services (except for emergency calls).
- The services offered to a roaming subscriber may be restricted by the capabilities of the visited network, and the roaming agreement between the visited and the home environment.

17.2 Principle

Long term evolution of the IM CN subsystem shall not be restricted by the short/mid-term inter-domain roaming requirements.

17.3 Requirements

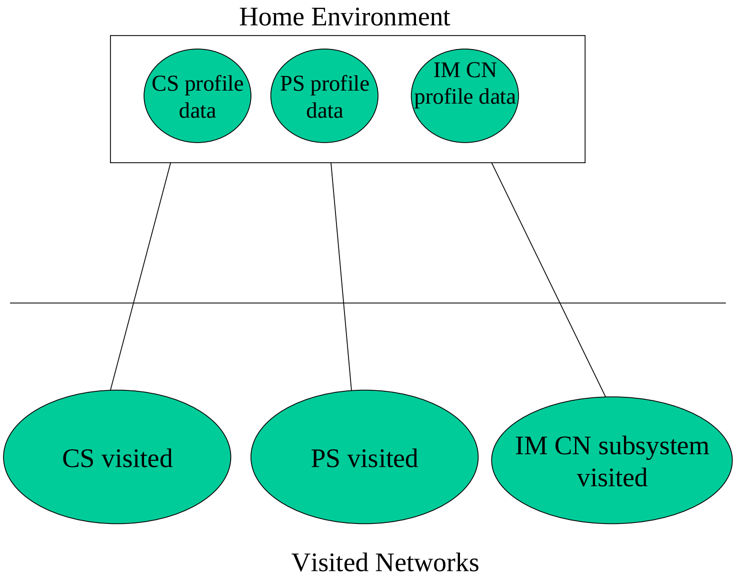

Figure 5: Roaming requirements

- The personalised services & capabilities available in a visited network are dependent upon the subscription options in the home environment. This does not preclude the visited network offering additional services, or access to content providers.
- Roaming from this release's home environment to CS (this release or earlier) visited network is required
- Roaming from this release's home environment to IM CN subsystem visited network is required
- Roaming from this release's home environment to PS (this release or earlier) visited network is required
- Roaming from previous releases' home environment (or earlier) to this release CS visited network is required
- Roaming from previous releases' home environment (or earlier) to this release PS visited network is required

Note: When an operator allows a subscriber to roam to different domains, the home environment needs to provide subscription data to the visited network . The mapping between service data of the different domains is not standardised; it is determined by the home environment and may be influenced by roaming agreements.

18 Handover Requirements

Any handover required to maintain an active service while a user is mobile within the coverage area of a given network, shall be seamless from the user's perspective. However handovers that occur between different radio environments may result in a change of the quality of service experienced by the user.

It shall be possible for users to be handed over between different networks subject to appropriate roaming/commercial agreements.

For further information see 3GPP TS 22.129 [9].

19 Network Selection

Network selection procedures are defined in 3GPP TS 22.011 [11].

Other procedures may be offered by the UE.

20 Security

Security matters are considered in 3GPP TS 21.133 [15] and 3GPP TS 33.120 [16].

21 Voice Call Continuity

21.1 General

The 3GPP system shall be able to provide continuity between CS voice services (Teleservice 11[14]) and the full duplex speech component of IMS multimedia telephony service [40] with no negative impact upon the user's experience of the voice service. This functionality is known as voice call continuity. Voice call continuity shall be executed when continuation of a voice service is required based on operator policy across a change in the connection of the UE to the 3GPP system as the user moves from using the CS domain to using IMS and vice versa.

The user experience shall be unaffected by the transition from a CS voice service to a full duplex speech component of IMS multimedia telephony and vice versa, and the user shall experience no disruption in the voice service provided. The voice service is continued with the same ME.

It shall be possible to support Voice call continuity between IMS and the CS domain belonging to different operators; i.e., when the user's IMS services are under the control of the home IMS and the user is roaming in the coverage of the visited CS network.

It shall be possible for an operator to enable or disable Voice call continuity for a given subscriber e.g. based on roaming conditions, terminal capabilities.

21.2 Support of Supplementary Services

The voice call continuity user's experience shall be such that, to the greatest degree possible, a consistency of service is provided regardless of the underlying communication infrastructure and technology. With regard to supplementary services, the general principle is that CS-based supplementary services only apply whilst a VCC subscriber is in the CS domain and equivalent services over IMS only apply whilst a VCC subscriber is in the IMS domain, although there are exceptions listed below. It is not required to synchronise the supplementary service settings of the CS domain with the related service settings of the IMS (e.g. different forwarding numbers may apply over CS and over IMS).

The following supplementary services apply. The impact on the supplementary services in case the VCC is executed for the calling party, the called party, or both is described below.

21.2.1 Line Identification Services

A user who has subscribed to the CLIP Supplementary Service and receives a call shall also receive the line identity or appropriate IMS information of the calling party.

The identity presentation is not changed for the duration of the call regardless of whether the call undergoes VCC.

If the CLIR Supplementary Service or IMS identity restriction is applicable to the call, then at call setup time the called user shall receive an indication that the identity is not available because of restriction.

The indication is not changed for the duration of the call regardless of whether the call undergoes VCC.

If COLP or a corresponding IMS service is applicable to a call the calling subscriber shall receive the connected line identity or appropriate IMS information at call setup time.

The identity presentation is not changed for the duration of the call regardless of whether the call undergoes VCC.

If the COLR Supplementary Service or IMS identity restriction is applicable to the call, then the calling user shall receive an indication at call setup time that the identity of the connected party is not available because of restriction.

The indication is not changed for the duration of the call regardless of whether the call undergoes VCC.

21.2.2 All Call Forwardings

It shall be possible to perform VCC on a call which was forwarded due to call forwarding supplementary services in the CS or redirecting services in the IMS.

21.2.3 Call Waiting

The functionality of call waiting supplementary service in the CS domain shall not be affected by the user's ability to undergo VCC.

Note: Whether the user will continue to receive call waiting notifications in the case their call is continued in the IMS will depend on whether a call waiting service is available in the IMS.

21.2.4 Call Hold

It shall be possible to re-establish a call which has been put on hold before undergoing VCC, after the VCC has been performed.

21.2.5 Multiparty

It shall be possible for any party in a multiparty call to undergo VCC and to stay in the call. It shall be possible to terminate the entire multiparty call when the served mobile subscriber releases even if she is connected via the IMS after undergoing VCC.

21.2.6 All Call Barrings

If a call were to undergo VCC and that would result in the call being barred in the target domain/system, it shall be up to the home operator policy whether the call continues in the target domain/system, the call terminates, or VCC is not executed for the call.

21.2.7 Void

21.2.8 Void

21.2.9 All other Supplementary Services

Other supplementary services are not discussed in this section as they do not apply to calls in progress (i.e. they apply to call set up only) or their support and/or the need for standardised implementation has not been identified as critical for VCC in this Release.

21.3 Quality of Service

Voice call continuity shall not adversely impact the quality of the voice service experienced by the user.

21.4 Security

Voice call continuity shall not adversely impact the security of the 3GPP system.

Security mechanisms of the 3GPP system shall be reused for voice call continuity.

21.5 Emergency calls

Voice call continuity for emergency calls shall be applicable to dual radio and single radio UEs.

Voice call continuity of emergency calls shall only be performed when all the following conditions are met:

- the source network is IMS;
- the target network supports emergency calls;
- the user is moving out of coverage;

- the source and target network belong to the same operator;
- the target network supports voice call continuity.

21.6 Charging

It shall be possible to indicate in the charging information that a VCC event has occurred (e.g., so that appropriate ratings can be applied for the CS and IMS parts of the continued voice call).

21.7 VCC Activation

It shall be possible to activate VCC based on operator policies, taking into account any of the following:

- Radio conditions e.g. radio quality thresholds and related hysteresis
- Coverage availability e.g.: Always prefer I-WLAN, if certain SSID are available

22 IMS Centralized Services

22.1 General

The ICS user shall receive both registered and unregistered services in a consistent manner when the user accesses IMS either via the CS or the PS domain (both of which can be supported by 3GPP access networks or non-3GPP access networks). Support of UEs enhanced with ICS capability as well as UEs without ICS capability shall be possible.

Note: The impacts to support non-3GPP CS domains to support of ICS are outside the scope of 3GPP.

22.2 Service Consistency

Subscribers shall have a consistent user experience regardless of the domain used, subject to the constraints of the UE and access network.

22.3 Service Continuity

ICS shall support service continuity between CS and PS domains (both of which can be supported by 3GPP access networks or non-3GPP access networks), subject to the constraints of the UE and access networks.

Note : The impacts to support non 3GPP CS domains are outside the scope of 3GPP.

The service continuity shall include:

- Basic services
- Non mid-call services
- Mid-call services

The support of service continuity for fax and data (CS) media components is not required.

22.4 IMS Services

The set of IMS services supported by ICS shall include at least the following, subject to the constraints of the UE and access networks:

- IMS Multimedia Telephony services, including:
  - speech,
  - video,
  - fax,
  - data (CS),
  - Supplementary services defined within [40]; and
  - Multimedia priority service.

Note: Other IMS hosted services supported by an operator or 3rd party may be supported.

ICS shall not limit the ICS user's capability to make emergency calls

22.5 Roaming Support

An ICS user shall be able to receive full ICS support from the HPLMN while roaming in the VPLMN, subject to the constraints of the VPLMN (e.g. roaming agreements, operator policies).

The Home operator shall be able to control if the UE enhanced with ICS capability shall act without ICS capability while roaming.

23 CS IP interconnection requirements

23.1 Introduction

CS IP interconnect represents the interconnection of MSC Server functionality between 2 CS networks over an underlying IP infrastructure.

23.2 IP interconnect

The IP connection used for CS IP interconnect shall be generic such that it can support all combinations of core network interconnection. E.g. the IP interconnection shall be shared between the IMS interconnection and the CS IP interconnection.

It shall be possible to handle the inter-connection of all services over this generic IP interface. The handling of security and charging shall also be generic for all IP interconnect scenarios.

23.3 MSC server interconnect

The following requirements apply at the interconnection point when two PLMNs are interconnected by means of IP transport technology for 2G and 3G CS services.

The system shall support the capability for CS service interoperability and interworking.

It shall be possible to apply operator defined policy at the interconnection point.

The system shall support the capability to control the session resources when two different network domains are connected that may have, for example, different IP addressing schemes.

The system shall support IP inter-connection between core networks either by direct connection or by using an intermediate carrier (e.g. GSMA IPX [43]).

The system shall support both bilateral interconnection between two carriers and multilateral interconnection (e.g. GSMA IPX [43]) by means of intermediate carrier.

The system shall support either

- transparent relay of the IP signalling and traffic;
- service aware interconnection

The system shall support codec negotiation across one or multiple interconnects to minimise transcoding (and preferably eliminate it) to provide the highest quality service to the user.

24 Service Alignment & Migration

24.1 Introduction

Services can be offered to the users via different service domains, e.g. certain teleservices and supplementary services via CS or Multimedia Telephony and supplementary services via IMS. Especially for the supplementary services given below a strong relationship exists from the user's point of view. Therefore means shall be provided to enable a consistent user experience when changing from one service domain to the other.

The requirements in this clause are applicable during the migration to an ICS. The ability to synchronise the service settings will facilitate the migration from a CS to an IMS based network and will allow network operators maximum flexibility in their migration strategies.

24.2 Alignment of supplementary services settings

Based on user preferences and if provisioned by the home operator the network shall automatically synchronise the parameter settings of the supplementary services listed in the following table:

CS Voice (TS11) Supplementary Services	Equivalent Multimedia Telephony Services in IMS domain	Service Behaviour Required
CLIP/CLIR	OIP/OIR	Consistency of presentation
CoLP/CoLR	TIP/TIR	Consistency of presentation
CNAP	OIP/OIR	Consistency of presentation
Call Forwarding	CDIV	Call forwarding/CDIV shall work consistently no matter which domain the user is in. The settings (e.g. forwarding numbers) shall remain the same across domains for all the parts of the service for which there is an equivalent.
Call Waiting	Communication Waiting	The busy state of the user shall be available to both domains so that this can be applied no matter from which domain an incoming call/communication originates. The activation status of Call Waiting and Communication Waiting shall be synchronised.
Call Hold	Communication HOLD	Any calls held in one domain shall remain held on moving to a different domain. It shall be possible to re-establish a call that was put on hold in another domain.
Multiparty	CONF	Any conference (multiparty) calls set up in one domain shall remain in force if any user moves to another domain.
Closed User Group	Closed User Group	Consistency required across domains
CCBS	CCBS	Consistency required across domains
Call Deflection	Defined in CDIV	Consistency required across domains
Explicit Call Transfer	ECommT	Consistency required across domains. The UE state (i.e. if busy or not) shall be available in both domains to ensure that this can be applied consistently.
Call Barring	Communication Barring	Call/ Communication Barring shall work consistently no matter which domain the user is in. The settings (i.e. barred numbers) shall remain the same across domains.
AoC	AoC	Consistent support across domains. If the user moves from one domain to the other during the communication, the AoC shall indicate the correct charge for the total duration of the communication.

The operator shall be able to provision the user with the possibility to define which of the above settings shall be synchronised automatically and which settings can exist independently of each other. E.g. a user might decide that the activation status of CLIP/OIP, CLIR/OIR, Call Waiting/Communication Waiting etc. is synchronised but that the call forwarding status and forwarded-to-number is different from the communication diversion settings and the diverted-to-party address.

If the synchronisation of supplementary services, which use the UE's busy state for invocation, is activated the busy state of the UE shall be available in both domains.

Note: The "user not reachable" or "user not logged in" conditions in the different domains are independent of each other. This means, e.g. not being registered in the IMS does not affect the invocation of CFNRc in the CS domain.

Synchronisation of settings means that the most recent changes which have been applied in one domain are propagated to the other domain.

There are certain circumstances under which the synchronisation will fail e.g. when a user inserts a SIP URI as diverted-to-party address in the IMS domain which has no Tel URI associated with it. In such a case there is no valid setting for the CS domain for this particular parameter and therefore the CS domain service setting shall remain unchanged. The user may receive a notification about the failure of the synchronisation procedure and its cause.

25 System optimisation for communication with specific characteristics

25.1 Void

25.2 Numbering Resource Efficiency

The following optional requirement is intended to provide better numbering resource efficiency for UEs that only require packet switched services.

- A network operator shall be able to provide PS only subscriptions with or without assigning an MSISDN.
- Remote triggering shall be supported with or without assigning an MSISDN.
- Remote UE configuration shall be supported without the use of an MSISDN.

Note 1: Current remote UE configuration solutions (i.e. Device Management and Over-the-Air configuration) are mainly based on SMS, which assumes the use of MSISDNs.

Note 2: The requirements in this sub-clause apply to server-to-UE communication scenarios only.

25.3 Network provided destination for uplink data

The Network Provided Destination for Uplink Data feature is intended for use when all data from a UE is to be directed to a network provided destination address.

- For uplink data communication, the network shall be able to direct all uplink PS data traffic to a network provided destination address.

Note: This feature may be used, for example, when all data from an electricity meter is sent to a server maintained by the network operator.

25.4 PS only subscriptions

The system shall support subscriptions that only allow packet based services and SMS.

26 Single Sign-On (SSO) Service

26.1 Requirements

26.1.1 Requirements for the UE

An SSO-capable UE shall support 3GPP SSO Authentication, without user intervention, based on Operator-controlled credentials.

An SSO-capable UE shall be able to initiate the SSO Service regardless of the access network technologies supported by the UE.

An SSO-capable UE that supports 3GPP access and non-3GPP access shall support transparency of the SSO Service from a user perspective during transitions between 3GPP access and non-3GPP access, whether or not the transition occurs during a data application session.

An SSO-capable UE may support a request for SSO Local User Authentication from a Data Application Provider or an Identity Provider to confirm the presence of the registered user of the data application.

26.1.2 Requirements for a 3GPP SSO Service

The 3GPP SSO Service shall provide secure, seamless and transparent access to data applications for users of the SSO Service independent of the access network technology.

The 3GPP SSO Service shall be able to interwork with Identity Management (IdM) specifications (e.g., OpenID [51]).

The 3GPP SSO Service shall support 3GPP SSO Authentication based on Operator-controlled credentials and policies.

The 3GPP SSO Service may support negotiation and use of an agreed authentication method between the UE and the 3GPP SSO Identity Provider. The negotiation of an authentication method may be repeated each time the user accesses a DAP's service.

The 3GPP SSO Service may support mechanisms to ensure the presence of the registered user of the data application to satisfy policies of the Data Application Provider.

The 3GPP SSO Service shall be transparent from a user perspective when transitions occur between 3GPP access and non-3GPP access, whether or not the transition occurs during a data application session.

The 3GPP SSO Service shall be transparent from a user perspective when the user accesses a data application using an identity created through a 3rd Party SSO Identity Provider. The user shall be able to configure which 3rd party SSO identities are used with the 3GPP SSO Service.

27 User plane congestion management

27.1 Introduction

RAN user plane congestion, in the context of this clause, is considered to be downlink congestion that affects the user plane, which may last for a few seconds, a few minutes, or a few hours due to arrival of new active users, increase of communication intensity of existing users, the radio environment changing, the mobile user changing location, and other reasons, thus causing the capacity of RAN resources to transfer user data to be exceeded. A short-duration burst of user plane traffic should not be identified as RAN congestion.

27.2 General

a) The network shall be able to detect RAN user plane congestion onset and abatement. Mechanisms to cope with RAN user plane congestions should be resilient to rapid changes in the level of congestion.

b) The network shall be able to identify whether or not an active UE is in a RAN user plane congested cell.

c) The network operator shall be able to configure or provision and enforce policy rules to best deal with RAN user plane congestion.

d) The system should react in a timely manner to manage a RAN user plane congestion situation, i.e. that the measures taken become effective to promptly help resolve the RAN user plane congestion.

e) The signalling overhead caused by RAN user plane congestion management solutions in the system shall be minimized.

f) The network shall be able to take into consideration the RAN user plane congestion status and the subscriber's profile when coping with traffic congestion.

27.3 Prioritizing traffic

a) According to operator policy, during RAN user plane congestion the operator shall be able to select the communications which require preferential treatment and allocate sufficient resources for such communications in order to provide these services with appropriate service quality.

b) According to operator policy, the network shall be able to select specific users (e.g. heavy users, roaming users, etc.) and adjust the QoS of existing connections/flows and apply relevant policies to new connections/flows depending on the RAN user plane congestion status and the subscriber's profile.

Note 1: Preferably, connections/flows need to be adjusted such that the user experience is not negatively affected.

27.4 Reducing traffic

a) Based on RAN congestion status and according to operator policy, the network shall be able to reduce the user plane traffic load (e.g. by compressing images or by adaptation for streaming applications) taking into account UE related information (e.g. UE capabilities, subscription).

b) The system shall be able to adjust the communication media parameters of real-time communications so that they consume less bandwidth.

27.5 Void

28 RAN Sharing Enhancements

28.1 General

RAN Sharing Enhancements allow multiple Participating Operators to share the resources of a single RAN according to agreed allocation schemes. The Shared RAN is provided by a Hosting RAN Operator which can be one of the Participating Operators.

The Shared RAN can be GERAN, UTRAN or E-UTRAN as described below.

All the following requirements shall be subject to Hosting RAN Operator configuration.

28.2 Specific E-UTRAN Sharing requirements

28.2.1 Allocation of Shared E-UTRAN resources

When E-UTRAN resources are shared they can be allocated unequally to the Participating Operators, depending on the planned or current needs of these operators and based on service agreements with the Hosting E-UTRAN Operator.

The following requirements apply:

The Hosting E-UTRAN Operator shall be able to specify the allocation of E-UTRAN resources to each of the Participating Operators by the following:

a) static allocation, i.e. guaranteeing a minimum allocation and limiting to a maximum allocation,

b) static allocation for a specified period of time and/or specific cells/sectors,

c) first UE come first UE served allocation.

Resources include both user plane and signalling plane. The Hosting E-UTRAN Operator needs to be able to manage the sharing of the signalling traffic independently from that of the user traffic because signalling traffic and user traffic are not always directly related. For example for MTC devices, the signalling traffic volume can be high and the user traffic volume low whereas for downloading video, the signalling traffic is low and the user traffic is high.

The following requirements apply:

The management and allocation of resources of signalling traffic over the Shared E-UTRAN shall be independent from the management and allocation of resources of the user traffic over the Shared E-UTRAN.

A Shared E-UTRAN shall be capable of differentiating traffic associated with individual Participating Operators.

A Shared E-UTRAN shall be able to conduct admission control based on the allocated E-UTRAN resources for each Participating Operator.

A Hosting E-UTRAN Operator shall be able to control resource usage taking into account the allocated E-UTRAN resources for each Participating Operator. A means of monitoring the usage of resources shall be provided.

All shared E-UTRAN capabilities offered by the Hosting E-UTRAN Operator shall be individually available for use by each Participating Operator where this is possible.

28.2.2 OA&M Access to the Shared E-UTRAN

Each Participating Operator can have their own OA&M capabilities, which are used for monitoring and for selected operations in a shared E-UTRAN. Information exchange involved in those operations need to be controlled by the Hosting E-UTRAN Operator as to prevent disclosing them to other Participating Operators, be it for business, operational, or technical reasons.

The following requirements apply:

Selected OA&M capabilities for the Shared E-UTRAN, under the control of the Hosting E- UTRAN Operator, shall be accessible by the Participating Operator's OA&M functions

This would allow, for example, the Participating Operator to do the following:

- test of communication path between the Participating Operator's network elements and the Shared E-UTRAN,
- obtain fault reports,
- retrieve RAN resource usage information.

28.2.3 Generation and retrieval of usage and accounting information

To facilitate interoperator accounting between Hosting E-UTRAN Operator and Participating Operator the Hosting E-UTRAN Operator needs to record E-UTRAN resource usage by UEs of the Participating Operator.

The following requirements apply:

A Hosting E-UTRAN Operator shall be able to collect events supporting the accounting of network resource usage separately for each Participating Operator. Collected events may be delivered to the subscriber's Participating Operator. This includes:

- start of service in the Shared E-UTRAN for a UE of the Participating Operator,
- end of service in the Shared E-UTRAN for a UE of the Participating Operator.

28.2.4 MDT Collection

MDT data from a Participating Operator's customer UEs allow the Hosting E-UTRAN Operator to be provided with performance measurements on his Shared E-UTRAN. The Participating Operator is responsible for obtaining and retaining any required user consent related to privacy.

The following requirements apply:

When authorized by the Participating Operators, the Hosting E-UTRAN Operator shall be able to collect MDT data of the Participating Operator's UEs connected through its E-UTRAN.

Note: This functionality should also allow for the case where the Hosting E-UTRAN Operator does not have an adjunct core network.

28.2.5 PWS support of Shared E-UTRAN

A Participating Operator potentially has regulatory obligations to initiate the broadcast of PWS messages regardless of E-UTRAN Sharing.

The following requirements apply:

The Shared E-UTRAN shall be able to broadcast PWS messages originated from the core networks of all Participating Operators.

Note: Rel-11 design requires a shared PWS core. However, some regulatory obligations require a solution in which no common PWS core network entity is involved.

28.2.6 Support for load balancing

Hosting E-UTRAN Operators have the need to optimize E-UTRAN resource usage within the shared E-UTRAN for a particular coverage area. At the same time, the agreed shares of E-UTRAN resources based on a single cell and sector for each Participating Operator need to be respected. Likewise, Participating Operators have the need to optimize their E-UTRAN resource usage among shared and unshared E-UTRAN for a particular coverage area.

The capability to perform load balancing on an individual Participating Operator's traffic basis within a shared E-UTRAN shall be supported.

The capability to perform load balancing on the combined traffic of all the Participating Operators within a shared E-UTRAN shall be supported.

The capability to perform load balancing between an individual Participating Operator's traffic within a shared E-UTRAN and traffic in that Participating Operator's unshared E-UTRAN where the shared and unshared E-UTRAN coverage overlaps shall be supported.

Note: Load balancing capabilities are expected to take into account the allocation of resources to each Participating Operator and the load level for each Participating Operator to the extent possible, so that the principal objective to maximize throughput is not impacted.

If load balancing in a Shared E-UTRAN is supported and if a Participating Operator’s EPC indicates overload to the Shared E-UTRAN in order to mitigate the overload situation then overload mitigation measures shall have minimal impact on the communication between the Shared E-UTRAN and other Participating Operators EPCs.

28.2.7 Dynamic capacity negotiation

In situations where a need for additional, unplanned, E-UTRAN capacity by a Participating Operator arises (e.g. in the case of big mass-events) the Shared E-UTRAN can provide means to allocate available spare capacity to the Participating Operator. Based on service level agreement between the Hosting and Participating operator such allocation can be automated without human intervention.

The following requirements apply:

The Participating Operator shall be able to query and request spare capacity of the Shared E-UTRAN, based on policies and without human intervention.

The Shared E-UTRAN shall be able to allocate spare capacity to Participating Operator, based on policies and without human intervention.

28.3 Specific GERAN & UTRAN Sharing Requirements

28.3.1 Allocation of Shared GERAN or UTRAN Resources

When GERAN or UTRAN resources are shared they can be allocated unequally to the Participating Operators, depending on the planned or current needs of these operators and based on service agreements with the Hosting RAN Operator.

The following requirements apply:

The Hosting RAN Operator shall be able to specify the allocation of GERAN or UTRAN resources to each of the Participating Operators by the following:

a) static allocation, i.e. guaranteeing a minimum allocation and limiting to a maximum allocation,

b) static allocation for a specified period of time and/or specific cells/sectors,

c) first UE come first UE served allocation.

The management and allocation of resources of signalling traffic over the Shared GERAN or UTRAN shall be independent from the management and allocation of resources of the user traffic over the Shared GERAN or UTRAN.

A Shared GERAN or UTRAN shall be capable of differentiating traffic associated with individual Participating Operators and shall be able to limit QoS available for traffic of the UEs of a Participating Operator (e.g. “best effort” if not enough resources are available).

A Shared GERAN or UTRAN shall be able to conduct admission control based on the allocated GERAN or UTRAN resources for each Participating Operator with a margin of tolerance.

A Hosting RAN Operator shall be able to control resource usage taking into account the allocated GERAN or UTRAN resources for each Participating Operator. A means of monitoring the usage of resources shall be provided.

All Shared GERAN or UTRAN capabilities offered by the Hosting RAN Operator shall be individually available for use by each Participating Operator where this is possible.

A Hosting RAN shall be able to provide an indication to each Participating Operator of how much capacity they are allocated at any one time. Signalling and User traffic shall be separately identified.

The ability to allow a single RAN (GERAN or UTRAN) to be fully shared by a number of Participating Operators shall be provided.

28.3.2 OA&M Access to the Shared GERAN or UTRAN

Each Participating Operator can have their own OA&M capabilities, which are used for monitoring and for selected operations in a Shared GERAN or UTRAN. Information exchange involved in those operations need to be controlled by the Hosting RAN Operator so as to prevent disclosing them to other Participating Operators, be it for business, operational, or technical reasons.

The following requirements apply:

Selected OA&M capabilities for the Shared GERAN or UTRAN, under the control of the Hosting RAN Operator, shall be accessible by the Participating Operator's OA&M functions

This would allow, for example, the Participating Operator to do the following:

- test of communication path between the Participating Operator's network elements and the Shared GERAN or UTRAN,
- obtain fault reports,
- retrieve GERAN or UTRAN resource usage information.

28.3.3 Generation and retrieval of usage and accounting information

To facilitate inter-operator accounting between the Hosting RAN Operator and the Participating Operator, the Hosting RAN Operator needs to record GERAN or UTRAN resource usage by UEs of the Participating Operator.

The following requirements apply:

A Hosting RAN Operator shall be able to collect events supporting the accounting of network resource usage separately for each Participating Operator. Collected events may be delivered to the subscriber's Participating Operator. This includes:

- start of service in the Shared GERAN or UTRAN for a UE of the Participating Operator,
- end of service in the Shared GERAN or UTRAN for a UE of the Participating Operator.

28.3.4 MDT Collection

MDT data from a Participating Operator's customer UEs allow the Hosting RAN Operator to be provided with performance measurements on his Shared GERAN or UTRAN. The Participating Operator is responsible for obtaining and retaining any required user consent related to privacy.

The following requirements apply:

When authorized by the Participating Operators, the Hosting RAN Operator shall be able to collect MDT data of the Participating Operator's UEs connected through its GERAN or UTRAN.

The Participating Operators shall be able to retrieve this data for their own subscribers from the Hosting RAN Operator.

Note: This functionality should also allow for the case where the Hosting RAN Operator does not have an adjunct core network.

28.3.5 PWS support of Shared GERAN or UTRAN

A Participating Operator potentially has regulatory obligations to initiate the broadcast of PWS messages regardless of GERAN or UTRAN Sharing.

The following requirements apply:

The Shared GERAN or UTRAN shall be able to broadcast PWS.

Note: The Hosting RAN Operator is responsible for the delivery of PWS messages to the UEs.

28.3.6 Support for load balancing

Hosting RAN Operators need to optimise GERAN or UTRAN resource usage within the Shared GERAN or UTRAN for a particular coverage area while respecting the agreed resource shares for each Participating Operator. Similarly, Participating Operators need to optimise their GERAN or UTRAN resource usage between Shared and unshared GERAN or UTRAN for a particular coverage area.

The Hosting RAN Operator shall have the capability to balance the Signalling and User Traffic load individually for each Participating Operator within a Shared GERAN or UTRAN.

The capability to perform load balancing on the combined traffic of all the Participating Operators within a Shared GERAN or UTRAN shall be provided. The agreed shares of GERAN or UTRAN resources shall be maintained. Signalling and User traffic shall be managed independently.

The capability to perform load balancing between an individual Participating Operator's traffic within a Shared GERAN or UTRAN and traffic in that Participating Operator's unshared GERAN or UTRAN where the Shared and unshared GERAN or UTRAN coverage overlaps shall be supported.

The Hosting RAN Operator shall be able to balance the load across all Participating Operators.

28.3.7 Dynamic capacity negotiation

The Participating Operator shall be able to request on-demand spare capacity of the Shared GERAN or UTRAN based on policies and without human intervention.

The Hosting RAN Operator shall be able to allocate spare on-demand capacity on the Shared GERAN or UTRAN to a Participating Operator, based on policies and without human intervention.

The Participating Operator shall be able to request the cancellation of granted on-demand capacity requests.

The Hosting RAN Operator shall be able to cancel a granted request based on a Participating Operator’s request or based on agreed policy.

The Hosting RAN Operator shall be able to change the amount shared by each Participating Operator based on traffic demand. A minimum capacity (that can be set) of the Hosting RAN shall be reserved for each Participating Operator that cannot be allocated to other Participating Operators.

29 Service exposure with 3rd party service providers

29.1 General

The intention of service exposure is that, under the assumption of a service agreement between MNO and a 3rd party, the 3GPP Network allows a 3rd party service provider to benefit from network provided services and capabilities that are exposed by the PLMN. For example the 3GPP Core Network can exchange information with the 3rd party to optimize usage and management of 3GPP resources. A standardized exposure of network services/capabilities reduces the complexity of different 3rd parties to access different 3GPP network services and capabilities.

General requirements for service exposure with 3rd party service providers:

- The operator shall be able to provide to a 3rd party service provider secure and chargeable access to the exposed services/capabilities i.e. to authenticate, authorize and charge the 3rd party entities.

NOTE 1: This requirement can be implemented by the existing standardised API frameworks e.g. the OMA API framework.

- It shall be ensured that the 3GPP services/capabilities are not disclosed to unauthorised parties and that user privacy (avoid e.g. trackable and traceable identity information of the concerned UE) is maintained subject to user agreement, operator policy, service agreement between operator and 3rd party and regulation constraints.
- The network service/capability exposure should be generic enough to support different application needs. Exposed 3GPP services/capabilities may use functionalities from different network entities and different 3GPP interfaces

29.2 Exposed Services and capabilities

The 3GPP Core Network shall be able provide a standardized interface to enable exposure of the following services and capabilities to 3rd party service providers:

Support of 3rd party interaction for 3GPP resource management for background data transfer:

- The 3GPP Core Network shall support a 3rd party service provider request for background data transfer to UEs that are served by the 3rd party service provider, indicating:
  - the desired time window for the data transfer,
  - the volume of the data expected to be transferred in a geographic area TS 23.032 [56].
- The 3GPP Core Network shall be able to inform the 3rd party service provider about:
  - one or more recommended time windows for the data transfer and
  - for each time window the maximum aggregated bitrate for the set of UEs in the geographical area indicated by the 3rd party service provider.
- Additionally, the 3GPP Core Network shall be able to inform the 3rd party service provider about the charging policy that will be applied to the 3rd party service provider if the data are transferred within the recommended time window and if transmission rates stay below the limits of the respective maximum aggregated bitrate.
- The goal of providing the time window is to favour transfer of more traffic during non-busy hours and reason for providing the maximum aggregate bitrate is to spread out traffic during that time. The goal of multiple time windows is to allow the 3rd party provider to choose one appropriate time window based on its preference like the expected charging regime and bitrate.

Support of 3rd party interaction on information for predictable communication patterns of a UE:

- The 3GPP Core Network shall enable a 3rd party service provider to provide information about predictable communication patterns of individual UEs or groups of UEs that are served by this 3rd party service provider. Such communication patterns may include:
  - Time and traffic volume related patterns (e.g. repeating communication initiation intervals, desired ‘keep alive’ time of data sessions, average/maximum volume per data transmission, etc.).
  - Location and Mobility related patterns (e.g. indication of stationary UEs, predictable trajectories of UEs, etc.).
- This information may be used by the 3GPP system to optimize resource usage.

Support of 3rd party requested session QoS and priority

- The 3GPP Core Network shall enable a 3rd party service provider to request setting up data sessions with specified QoS (e.g. low latency or jitter) and priority handling to a UE that is served by the 3rd party service provider.

Support of 3rd party requested broadcast

- The 3GPP Core Network shall enable a 3rd party service provider to request sending a broadcast message in a specified geographic area (as specified in TS 22.368 [52]) expecting to reach a group of devices that are served by the 3rd party service provider.

Informing the 3rd party about potential network issues

- The 3GPP Core Network shall be able to indicate to a 3rd party service provider when data transmissions have a risk of incapability to provide expected throughput and/or QoS in a specific area (e.g. due to forecasted high traffic load in that area). Additionally, an estimate may be given when the high traffic load is expected to be mitigated.

Informing the 3rd party about UE status

- The 3GPP Core Network shall be able to provide the following information about a UE that is served by the 3rd party service provider:
  - Indication of the of the roaming status (i.e. Roaming and No Roaming) and the serving network, when the UE starts/stops roaming,
  - Loss of connectivity of the UE,
  - Change or loss of the association between the ME and the USIM,
  - Communication failure events of the UE visible to the network (e.g. for troubleshooting).
  - Reporting when the UE moves in/out of a geographic area that is indicated by the 3rd party,
  - Reporting when the UE changes Routing Area / Tracking Area / Location Area / Cell.

Note: The area indicated by a 3rd party service provider can be mapped to the area used in the 3GPP network, i.e. a list of LAs/RAs/TAs. The 3rd party service provider can define a geographical area as shapes (e.g. polygons, circles) or civic addresses (streets, districts…) as referenced by OMA Presence API [53] e.g. defined by shape areas of IETF RFC-5491 [54] or by civic addresses defined in IETF RFC-5139 [55].

- The 3rd party service provider shall be able to request a one time reporting or reporting at regular times on the number of UEs present in a certain area and the location of each UE as for a Location Based Service.

Informing the 3rd party about a UE's connection properties

- The 3GPP Core Network shall be able to inform a 3rd party about a UE's connection properties.

Note: Connection properties of a UE describe the average data rate range or non-absolute value (e.g. high, medium or low) that the UE is likely to be able to obtain at the current location. The connection properties can, for example, be generated from the UE's RAT type the UE is currently attached to, the load conditions at its current location and/or other parameters.

Support for non-IP small data transfer with a 3rd party

- The 3GPP Core Network shall support a 3rd party for submitting a small amount of non-IP data for delivery to a UE.
- The 3GPP Core Network shall support a 3rd party application server for receiving a small amount of non-IP data delivered from a UE.
- The 3GPP Core Network shall support a 3rd party to configuring non-IP data delivery for a particular UE (e.g. destination address, maximum number of messages, duration for which configuration applies).

Note: The use of the Non-IP Data Delivery feature via Service Capability Exposure Function assumes that the UE has indicated support for 'non-IP data transfer' and in case the Service Capability Exposure Function is used there will be no user plane EPS bearer established.

30 Flexible Mobile Service Steering

30.1 Introduction

In order to realize efficient and flexible mobile service steering in the operator deployed (S)Gi-LAN, the network operator uses information (e.g. user profile, network operator's policies, RAT type, application characteristics) to define traffic steering policies. These policies are used to steer the subscriber's traffic to appropriate enablers (e.g. NAT, antimalware, parental control, DDoS protection) in the (S)Gi-LAN.

The term (S)Gi-LAN used in the present document represents a system which is out of 3GPP scope.

30.2 General Requirements

The following requirements apply:

- The 3GPP Core Network shall be able to define and modify traffic steering policies that are used to steer traffic in (S)Gi-LAN e.g. to improve e.g. the user's QoE, apply the bandwidth limitation and provide valued added services.
- Traffic steering policy shall be able to distinguish between upstream and downstream traffic.
- The 3GPP Core Network shall be able to define different traffic steering policies for a user's traffic on a per session basis (e.g. for applying parental control, anti-malware, DDoS protection, video optimization).
- The 3GPP Core Network defines traffic steering policies based on one or more pieces of information such as:
  - network operator's policies
  - user subscription (e.g. user's priority, the status of optional subscriber services from the subscription data, based on service provider used, subscribed QoS)
  - user's current RAT
  - the network (RAN and CN) load status
  - application characteristics such as: application type (video, web browsing, IM, etc), application protocol (HTTP, P2P, etc), target address name (URL) and application provider (My tube, etc)
  - time
  - UE location
  - information (e.g. APN) of the destination network (i.e. a PDN or an internal IP network) of traffic flow
- In case of roaming, the HPLMN shall be able to apply traffic steering policies for home routed traffic.

Note: Traffic steering policies for local breakout is not specified.

Annex A (normative): Description of optional user equipment features

A.1 Display of called number

This feature enables the caller to check before call setup whether the selected number is correct.

A.2 Indication of call progress signals

Indications shall be given such as tones, recorded messages or visual display based on signalling information returned from the PLMN. On data calls, this information may be signalled to the DTE.

Call progress indicators are described in 3GPP TS 22.001 [4].

A.3 Country/PLMN indication

The country/PLMN indicator shows in which PLMN the UE is currently registered. This indicator is necessary so that the user knows when "roaming" is taking place and that the choice of PLMN is correct. Both the country and PLMN will be indicated. When more than one visited PLMN is available in a given area such information will be indicated.

The PLMN name is either:

- Stored in the ME and associated with the MCC+MNC combination received on the broadcast channel;
- NITZ (see 22.042 [17]) (in which case it overrides the name stored in the UE);
- stored in the USIM in text and /or graphic format and associated with the MCC+MNC combination, and optionally the LAI, received on the broadcast channel (in which case it overrides the name stored in the UE and – if present – the NITZ name).

It shall be possible to store on the USIM at least 10 PLMN Identifications (MCC+MNC combination and optionally the LAI) for which the same PLMN name shall be displayed.

The PLMN name stored in the USIM has the highest priority, followed by the PLMN name provided by NITZ. The PLMN name stored in the ME has the lowest priority.

If the PLMN name stored in the USIM is not available in text format and the UE is unable to display the graphic format, the PLMN name provided by NITZ has the highest priority, the PLMN name stored in the ME has the next priority.

A.4 Service Provider Name indication

The service provider name is stored in the USIM in text and/or optionally graphic format. It shall be possible to associate at least 10 PLMN Identifications (MCC+MNC combination) with the same SP Name.

When registered on the HPLMN, or one of the PLMN Identifications used for Service Provider Name display:

(i) The SP Name shall be displayed;

(ii) Display of the PLMN Name is an operator's option by setting the appropriate fields in the USIM (i.e. the Service Provider name shall be displayed either in parallel to the PLMN Name or instead of the PLMN Name).

When registered on neither the HPLMN, nor one of the PLMN Identifications used for Service Provider Name display:

(i) The PLMN name shall be displayed;

(ii) Display of the SP Name is an operator's option by setting the appropriate fields in the USIM.

If the UE is unable to display the full name of the Service Provider the name is cut from the tail end. The storage of Service Provider name and options, and choice of options, shall be under control of the network operator.

A.4a Core Network Operator Name indication

It shall be possible for the UE to display the name of the core network operator the user has selected.

A.5 Keypad

A physical means of entering numbers, generally, though not necessarily, in accordance with the layout shown in figure A.1.

See also 3GPP TS 22.030 [6] (Man-Machine Interface).

Additional keys may provide the means to control the UE (e.g. to initiate and terminate calls).

1	**2**	**3**
4	**5**	**6**
7	**8**	**9**
*****	**0**	**#**

Figure A.1

A.6 Short message indication and acknowledgement

This feature allows the delivery of short messages to a UE from a service centre. Such messages are submitted to the service centre by a telecommunications network user who can also request information of the status of the message by further interrogation of the service centre. The service centre then transmits the message to an active UE user.

The UE must therefore provide an indication to the user that a message has been received from the service centre and must also send an acknowledgement signal to the PLMN to show that this indication has been activated. The PLMN then returns this acknowledgement to the service centre.

The short message service teleservice is described in specification 3GPP TS 22.003 [14].

A.7 Short message overflow indication

An indication shall be given to the user of the short message service when an incoming message cannot be received due to insufficient available memory.

A.8 International access function

Provision is made for a direct, standard method of gaining international access. For this purpose the UE may have a key whose primary or secondary function is marked "+". This is signalled over the air interface and would have the effect of generating the international access code in the network. It may be used directly when setting up a call, or entered into the memory for abbreviated dialling.

This feature is of benefit since the international access code varies between CEPT countries, which might cause confusion to a user, and prevent the effective use of abbreviated dialling when roaming internationally. Users may still place international calls conventionally, using the appropriate international access code.

A.9 Service Indicator (SI)

An indication is given to the user that there is adequate signal strength (as far as can be judged from the received signal) to allow a call to be made. Additionally the network should be capable of providing, and the UE may display, an indication of the serving cells' capabilities e.g. GPRS, HSDPA, HSUPA.

If indicated by the registered network, the UE in idle mode may display an indication of one of the radio access technologies provided to the UE in the network on which the UE is registered with the following priority order: E-UTRAN, UTRAN, GERAN.

Displaying the serving cell's capabilities and the access technology are mutually exclusive.

A.10 Dual Tone Multi Frequency (DTMF)

The UE shall be capable of initiating DTMF in accordance with specifications 3GPP TS 22.003 [14]. Optionally, the UE may provide a suppress function which allows the user to switch off the DTMF function.

A.11 On/Off switch

The UE may be provided with a means of switching its power supply on and off. Switch-off shall be "soft", so that on activation, the UE completes the following housekeeping functions: termination of a current call, detach (where applicable) and storing required data in the SIM/USIM before actually switching off. As far as possible, this procedure should also apply on power failure (e.g. remote switch-off or low battery).

A.12 Sub-Address

This feature allows the mobile to append and/or receive a sub-address to a Directory Number, for use in call set-up, and in those supplementary services that use a Directory Number.

A.13 Short Message Service Cell Broadcast

The Short Message Service Cell Broadcast enables the mobile equipment to receive short messages from a message handling system.

The short message service cell broadcast teleservice is described in specification 3GPP TS 22.003 [14]

A.14 Short Message Service Cell Broadcast DRX

This feature enables a mobile equipment to save on battery utilization, by allowing the mobile equipment to not listen during the broadcast of messages the subscriber is not interested in.

A.15 Support of the extended Short message cell broadcast channel

This feature allows a mobile equipment by supporting of the extended Short message cell broadcast channel to enhance the capacity of the service. The support of the extended channel has low priority, i.e. the UE can interrupt the reading of this channel if idle mode procedures have to be executed.

A.16 Network Identity and Timezone

The feature provides the means for serving PLMNs to transfer current identity, universal time and the local timezone to mobile equipments, and for the mobile equipments to store and use this information. This enhances roaming by permitting accurate indication of PLMN identities that are either newer than the ME or have changed their name since the ME was sold. Additionally time and timezone information can be utilized by MEs as desired.

The network name time and timezone information will normally be transferred from the network to the ME:

1) Upon registering on the network.

2) When the UE geographically relocates to a different Local Time Zone.

3) When the network changes its Local Time Zone, e.g. between summer and winter time.

4) When the network changes its identity.

5) At any time during a signalling connection with mobile equipment.

Further details of this feature are described in 3GPP TS 22.042 [15].

A.17 Network's indication of alerting in the UE

This feature provides the means for serving PLMNs to transfer to a UE an indication that may be used by the UE to alert the user in a specific manner in the following cases:

- mobile terminating call
- network initiated USSD
- network initiated Mobile Originated (MO) connection, if the ME supports the "network initiated MO connection " feature.

Eight different indications are defined, whether the mobile terminating traffic is a call or USSD or related to the network initiated MO connection procedure. These indications are sent by the network and received by the UE:

- Three of these indications are used as levels, reflecting some kind of urgency: level 0 indicates that the UE shall not alert the user for USSD and remain silent in the case of call, level 2 shall be considered by the UE as more important than level 1 for the purpose of alerting the user.
- The five other indications are used as categories, identifying different types of terminating traffic. The UE shall inform the user in a specific manner for each of these five categories. Nevertheless, the possible forms of the alert (different ringing tones, displayed text, graphical symbols...) is still up to the mobile manufacturer (some forms of alerts can be simultaneously used, e.g. ringing tones and text on the display).

The management of the feature by the UE requires for the handling of categories that :

- the SIM/USIM stores for each category an informative text (maximum 25 characters per category) describing the type of terminating traffic associated with the category. This information could be used by the UE when alerting the user (display on the screen). It is necessary for the network operator to be able to change the meaning of each category.
- The user has the ability to set up his/her own association between the type of terminating traffic (identified by each category) and the different types of alert provided by the UE. To help the user in this choice, the UE uses the informative text associated with each category (as stored in the SIM/USIM). The UE should keep this association when switched off.

  Default settings should also be defined in the ME for the following cases :

  - when the UE receives a call, USSD or a request for a network initiated MO connection with no alerting indication,
  - when the UE receives a call, USSD or a request for a network initiated MO connection with a category of alerting not defined in the SIM/USIM.

  These default settings should be separated per type of mobile terminated traffic received (call, USSD or request for a network initiated MO connection).

A UE supporting the feature shall act according to the following points in case of mobile terminating traffic :

- when a mobile terminating traffic is received without any indication (level or category), the ME shall act as if it was not supporting the feature, i.e. use a default alert (e.g. associated with this type of mobile terminating traffic).
- if a level is indicated, the UE shall use an alert enabling the user to differentiate between the three levels.
- if a category is indicated, then :
    - if the SIM/USIM used in the UE does not store any information on that feature, the UE shall ignore the category received with any mobile terminating traffic and act as if it was not supporting the feature, i.e. use a default alert (e.g. associated with this type of mobile terminating traffic).
    - if the category is not defined in the SIM/USIM, the UE shall act as if it was not supporting the feature, i.e. use a default alert (e.g. associated with this type of mobile terminating traffic).
    - if the category is defined in the SIM/USIM, the UE shall use the alert associated with this category. In addition, it would be very useful for the user to be notified of the informative text associated with this category (e.g. on the display).

Some interactions between this feature and other services related to alerting are described below :

- the call waiting service has priority on this feature, i.e. the call waiting tone will be played and not the alert derived by this feature. If possible, two different indications should be given to the user (e.g. the call waiting tone and a text on the display indicating call waiting, and in addition a text relative to the type of the new call received).
- the presentation of the calling line identity takes priority on this feature, if it is not possible to display this information and another information related to this feature.
- In case of interaction between this feature and UE specific features to alert the user (e.g. whole silent mode), the user should still be able to differentiate between the different levels or different types of terminating traffic, even if the alert itself may be changed.

A.18 Network initiated Mobile Originated (MO) connection

The "Network Initiated Mobile Originated connection" feature allows the network to ask the mobile equipment to establish a mobile originated connection. The serving PLMN provides the mobile equipment with the necessary information which is used by the mobile equipment to establish the connection.

Currently only the network initiated mobile originated call feature is specified. It is mandatory for a UE supporting CCBS and is used in the case of a CCBS recall.

A.19 Abbreviated dialling

The directory number or part of it is stored in the mobile equipment together with the abbreviated address. After retrieval the directory number may appear on the display.

Abbreviated dialling numbers stored in the UE or USIM may contain wild characters.

If wild characters are used to indicate missing digits, each wild character shall be replaced for network access or supplementary service operation, by a single digit entered at the keypad. The completed directory number is transmitted on the radio path.

A.20 Barring of Dialled Numbers

This feature provides a mechanism so that by the use of an electronic lock it is possible to place a bar on calling any numbers belonging to a pre-programmed list of numbers in the SIM/USIM.

Barred Dialling Numbers stored in the /USIM may contain wild characters.

Under control of PIN2, "Barred Dialling Mode" may be enabled or disabled. The selected mode is stored in the SIM/USIM.

Under PIN2 control, it shall be possible to add, modify or delete a particular "Barred Dialling Number" (BDN) and to allocate or modify its associated comparison method(s). This BDN may have the function of an abbreviated dialling number / supplementary service control (ADN/SSC), overflow and/or sub-address.

When BDN is inactive, no special controls are specified, and the barred dialling numbers may be read (though not modified or deleted, except under PIN2 control) as if they were normal abbreviated dialling numbers. Access to keyboard and normal abbreviated dialling numbers (including sub-address) is also permitted.

When Barring of Dialled Numbers is active:

- Considering a number dialled by the user, if it exists a BDN for which there is a successful comparison (see below) between that BDN and the dialled number, then the ME shall prevent the call attempt to that number. If there is no BDN to fulfil those conditions, the call attempt is allowed by the ME.

With each BDN is associated one (or a combination of) comparison method(s) used between that BDN and the number dialled by the user. At least three different comparison methods are possible:

- The comparison is made from the first digit of that BDN, from the first digit of the dialled number and for a number of digits corresponding to the length of the BDN.

- The comparison is made from the first digit of that BDN, from any digit of the dialled number and for a number of digits corresponding to the length of the BDN.

- The comparison is made backwards from the last digit of that BDN, from the last digit of the dialled number and for a number of digits corresponding to the length of the BDN.

- If a BDN stored in the SIM/USIM contains one or more wild characters in any position, each wild character shall be replaced by any single digit when the comparison between that BDN and the dialled number is performed.

- If a BDN contains a sub-address, and the same number without any sub-address or with that sub-address is dialled, the ME shall prevent the call attempt to that number.

- Numbers specified as "barred" may only be modified under PIN2 control.

- If the ME does not support barring of dialled numbers, the UE with a BDN enabled USIM shall not allow the making of calls and the UE with BDN enabled SIM shall not allow the making or receiving of calls. However, this feature does not affect the ability to make emergency calls.

If "Fixed Number Dialling" and "Barring of Dialled Numbers" are simultaneously active, the dialled number shall be checked against the two features before the ME allows the call attempt. In that case, a dialled number will only be allowed by the ME if it is in the FDN list and if the comparison between that number and any number from the BDN list is not successful.

The UE may support other selective barrings, e.g. applying to individual services (e.g. telephony, data transmission) or individual call types (e.g. long distance, international calls).

A.21 DTMF control digits separator

Provision has been made to enter DTMF digits with a telephone number, and upon the called party answering the UE shall send the DTMF digits automatically to the network after a delay of 3 seconds (± 20 %). The digits shall be sent according to the procedures and timing specified in 3GPP TS 24.008 [13].

The first occurrence of the "DTMF Control Digits Separator" shall be used by the ME to distinguish between the addressing digits (i.e. the phone number) and the DTMF digits. Upon subsequent occurrences of the separator, the UE shall pause again for 3 seconds (± 20 %) before sending any further DTMF digits.

To enable the separator to be stored in the address field of an Abbreviated Dialling Number record in the SIM/USIM, the separator shall be coded as defined in 3GPP TS 31.102 [19]. The telephone number shall always precede the DTMF digits when stored in the SIM/USIM.

The way in which the separator is entered and display in the UE, is left to the individual manufacturer's MMI.

MEs which do not support this feature and encounter this separator in an ADN record of the SIM/USIM will treat the character as "corrupt data" and act accordingly.

A.22 Selection of directory number in messages

The Short Message Service (SMS), Cell Broadcast Service (CBS), Multimedia Message Service (MMS), Network Initiated USSD or Network Response to Mobile Originated USSD message strings may be used to convey a Directory Number, which the user may wish to call.

A.23 Last Numbers Dialled (LND)

The Last "N" Numbers dialled may be stored in the SIM/USIM and/or the ME. "N" may take the value up to 10 in the SIM/USIM. It may be any value in the ME. The method of presentation of these to the user for setting up a call is the responsibility of the UE but if these numbers are stored in both the SIM/USIM and the UE, those from the SIM/USIM shall take precedence.

A.24 Service Dialling Numbers

The Service Dialling Numbers feature allows for the storage of numbers related to services offered by the network operator/service provider in the SIM/USIM (e.g. customer care). The user can use these telephone numbers to make outgoing calls, but the access for updating of the numbers shall be under the control of the operator.

Note: No MMI is envisaged to be specified for these numbers and it is left to mobile manufacturer implementations.

A specific example of Service Dialling Numbers is the storage of mailbox dialling numbers on the SIM/USIM for access to mailboxes associated with Voicemail, Fax, Electronic Mail and Other messages.

A.25 Fixed number dialling

This feature provides a mechanism so that by the use of an electronic lock it is possible to place a bar on calling any numbers other than those pre-programmed in the SIM/USIM.

Under control of PIN 2, "Fixed Dialling Mode" may be enabled or disabled. The mode selected is stored in the SIM/USIM.

Fixed Dialling Numbers (FDNs) are stored in the SIM/USIM in the Fixed Dialling Number field. FDN entries are composed of a destination address/Supplementary Service Control. Destination addresses may have the format relevant to the bearer services/teleservices defined in [21] and [14]. FDN entries may take the function of an Abbreviated Dialling Number/Supplementary Service Control (ADN/SSC), Overflow and/or sub-address. Fixed Dialling Numbers stored in the SIM/USIM may contain wild card characters.

The Fixed Dialling feature is optional, however when Fixed Dialling Mode is enabled, an ME supporting the feature shall;

- Prevent the establishment of bearer services/teleservices to destination addresses which are not in FDN entries on a per bearer service/teleservice basis. The list of bearer services/teleservices excluded from the FDN check shall be stored in the SIM/USIM. Those bearer services/teleservices are characterised by their service code as described in [23]. For instance if the SMS teleservices is indicated in this list, SMS can be sent to any destination. By default, the ME shall prevent the establishment of any bearer service/teleservice to destination addresses which are not in FDN entries.

- Only allow modification, addition or deletion of Fixed Number Dialling entries under control of PIN2.

- Allow the establishment of bearer services/teleservice to destination addresses stored in FDN entries. For SMS, the Service Center address and the end-destination address shall be checked.

- Support the reading and substitution of wildcards in any position of an FDN entry, via the ME MMI.

- Allow the user to replace each wildcard of an FDN entry by a single digit, on a per call basis without using PIN2. The digit replacing the wildcard may be used for network access or supplementary service operation.

- Only allow Supplementary Service (SS) Control (in Dedicated or Idle mode) if the SS control string is stored as an FDN entry.

- Allow the extension of an FDN entry by adding digits to the Fixed Dialling number on a per call basis.
- Allow the emergency numbers (see Section 8.4) to be called, even if it is not an FDN entry.
- Allow normal access to ADN fields (i.e. allow ADN entries to be modified, added or deleted) and the keyboard.
- Allow use of ADNs subject to the FDN filter.

When FDN is disabled, an ME supporting FDN shall;

- Allow FDN entries to be read as though they were normal ADN entries.
- Only allow modification, addition or deletion of Fixed Number Dialling entries under control of PIN2.
- Allow normal access to ADN fields and the keyboard.

If the ME does not support FDN, the UE shall not allow the making of calls when Fixed Dialling is enabled in the USIM, and the UE shall not allow the making or receiving of calls when Fixed Dialling is enabled in the SIM. However, emergency calls (112 and other user defined emergency numbers) shall still be possible.

Note: Wildcards are stored on the SIM/USIM. The wildcard coding is given in 3GPP TS 31.102 [19].

A.26 Message Waiting Indication

A short message may be used to provide an indication to the user about the status and number of types of messages waiting on systems connected to the PLMN. The ME shall present this indication as an icon on the screen, or other MMI indication, and store the indication status on the SIM/USIM to allow the status to be retained through power off/on, SIM/USIM movement between UEs etc..

The ME shall be able to accept and acknowledge these message waiting status short messages irrespective of the memory available in the SIM/USIM.

A.27 Requirements for the transfer of eCall Minimum Set of Data (MSD)

With the exception of the following specific requirements, considered necessary for the satisfactory operation of the eCall service, all existing TS12 emergency call requirements shall apply.

An eCall shall consist of a TS12 emergency call supplemented by a minimum set of emergency related data (MSD). The MSD e.g. vehicle identity, location information and other parameters, is defined by CEN [46].

- An eCall may be initiated automatically, for example due to a vehicle collision, or manually by the vehicle occupants;
- An IVS, or other UE designed to support eCall functionality, shall include in the emergency call set-up an indication that the present call is either a Manually Initiated eCall (MIeC) or an Automatically Initiated eCall (AIeC).
- The Minimum Set of Data (MSD) sent by the In vehicle System (IVS) to the network shall not exceed 140 bytes;
- The MSD should typically be made available to the PSAP within 4 seconds measured from the time when end to end connection with the PSAP is established;
- Should the MSD component not be included in an eCall, or is corrupted or lost for any reason, then this shall not affect the associated TS12 emergency call speech functionality.
- A call progress indication shall be provided to the user whilst the MSD transmission is in progress.
- To reduce the time taken to establish an eCall an IVS whilst in eCall only mode, may receive network availability information whilst not registered on a PLMN.
- PLMNs should make use of eCall indicators, received in the emergency call set-up, to differentiate eCalls from other TS12 emergency calls.

- The MIeC and AIeC should be used by the serving network to filter and route eCalls to dedicated, eCall equipped, PSAPs.
- Where the eCall indicators are not supported by the serving network, the time needed for the PSAP eCall modem to differentiate between eCalls and other TS12 calls, before routing the call to an operator, shall not exceed 2 seconds from when the IVS receives notification that the PSAP has answered the call.
- The PSAP shall be given an indication that the incoming call is an eCall.
- When a TS12 emergency call, other than an eCall, is routed to a PSAP that supports the eCall service, the eCall equipment shall not cause audible interference to the emergency voice call.
- When an eCall is routed to a PSAP, that does not support the eCall service, the In Vehicle System (IVS) shall not emit any audible tones for longer than 2 seconds from the time that the call is first answered.

Throughout the duration of the emergency call and following receipt of the MSD by the PSAP

- It shall be possible for the PSAP to send a confirmation to the IVS that the MSD has been acted upon.
- It shall be possible for the PSAP to request the IVS to re-send its most recent MSD.
- It shall be possible for the PSAP to instruct the IVS to terminate the eCall.

A.28 Requirements for "In Case of Emergency" (ICE) information

The In Case of Emergency (ICE) information are used to enable first responders, such as paramedics, fire-fighters, and police officers, to contact a victim's emergency contact(s) in order to identify the victim and obtain important medical information or other information required during an emergency.

A UE shall have the capability to store one or more "ICE information" on the UICC. The "ICE information" shall list the type of information which is to be configured by the mobile operator as described in the table below.

Table A.28.1: "ICE information" description

	ICE information type format	ICE information type value	ICE information value 1	ICE information value 2	ICE information value n
Description	Two formats shall be defined in this release: 1- Phone number format. For the phone number format it shall be possible to store a name and a phone number. 2- Free format. It shall be possible for the mobile operator to provision zero, one or several instances of a given format on the UICC.	It shall be possible to store this information in text or graphic format.	It shall be possible to have at least one ICE information value field. If more than one information value fields are required it shall be indicated by the ICE information type format. It shall be possible for the subscriber to add, modify, view, or delete any ICE information value. For the free format ICE information type format, it shall be possible to store	Present only if explicitly indicated by the ICE information type format.	Present only if explicitly indicated by the ICE information type format.

			information in text or graphic format. For pre-defined ICE information type formats the ME should adapt the input mode to the type of the ICE information value (e.g. numerical mode for phone number input).		

The following table provides an example of ICE information stored on the UICC:

Table A.28.2: "ICE information" example

ICE information type	ICE information type value	ICE information value 1	ICE information value 2
Phone Number	"Contact in case of emergency"	My Wife	+3364566
Phone Number	"Contact in case of emergency"	Family Smith	+3364565
Phone Number	"Contact in case of emergency"	My Family doctor: Dr. Jones	+33643234
Free Format	"Medical Information"	My blood type is A+, I am allergic to etc.	N/A
Free Format	"Home Postal Address"	15 rue de la Paix, Paris, France	N/A
Free Format	"Language"	French	N/A
Free Format	"Travel Information"	London, from 3rd July. to 29th July, 2008	N/A

Provision is made for direct and unambiguous read access from the UE to ICE information stored on the UICC.

The subscriber may choose not to enter any ICE information.

The default configuration for this information shall be that they are accessible even when the security features on either the UE or UICC have been enabled (e.g. the keypad is locked). It shall be possible for the subscriber to change this default setting to prevent accessibility to the ICE information (i.e. a user configurable setting in the UICC indicates whether the ICE information on the UICC shall be displayed or not, if the ICE access procedure described in TS 22.030 [6] is invoked).

The unlocking of ICE information shall not allow access to other secure information on the UICC or UE. The ICE information value should not be accessible to ME or UICC applications.

The ICE access procedure is described in TS 22.030 [6].

Annex B (informative): Additional number use case

Company X is customer of operator Z and has a subscription for voice and data with 10 SIM cards. The company also decides to subscribe to the private numbering plan feature of operator Z. A certain corporate phone number is assigned to company X, e.g. +43 676 88888 xx (with xx being the extensions of the users), and they can freely choose the extensions for up to 100 SIM cards.

Initially company X take the existing 10 SIM cards (with the existing MSISDNs) and integrate them into the private numbering plan by assigning the extensions 11 to 20 to these cards via a web portal. This is not a permanent assignment, the extensions can be changed easily any time, old SIMs can be deleted from the private numbering plan and new ones can be integrated whenever necessary. Special privileges or restrictions can be chosen on a per SIM basis via the portal and special tariffs apply.

The CEO of company X calls a customer. The customer's phone displays the corporate number + extension of the CEO as CLI. After a while the customer decides to call back the CEO and uses the corporate number + extension as MSISDN. Later they also exchange some short messages, again the corporate number + extension of the CEO is used as sender's MSISDN.

For all these calls and short messages operator Z is able to collect charging records containing the corporate number + extension.

The CEO also uses her smart phone to establish PS connections to browse the internet and to access the intranet via the operator's gateway. The corporate number + extension is used for authentication and authorisation in the company's intranet and its web applications. Charging for PS connections by operator Z is also based on this number.

In general the users in company X are not aware of their "real" underlying MSISDNs in the system, they only see the corporate number + their extension and there is no need for them to know the underlying MSISDN.

Annex C (informative): Change history

Change history					
SMG No.	**TDoc. No.**	**CR. No.**	**Section affected**	**New version**	**Subject/Comments**
SMG#22	302/97	001	4.6 (Role Model)	3.1.0	SMG3 queried the separation of network operator into core and access, which, on examination, SMG1 find unhelpful
SMG#22	319/97 (SMG1 WPC 125/97)	002		3.1.0	Editorial Changes: FLMPTS was replaced by IMT 2000, 2 new references given, additional clarifications.
SMG#22	320/97	003	8.5, 9.3, 9.5, 17	3.1.0	Changes on Emergency Calls, User identification, Multiple profiles and additional handover requirements.
After SMG#23	SMG1 433u/97 965/97	004		Draft 3.2.0	Based on Approved Changes at SMG#22 Distributed at SMG1 in Dresden Nov 3-7, 97 to be Approved at SMG#24
SMG#24	966/97	005	Sections 8, 9, 11	3.2.1	Restructuring of sections 8,9 and 11 to gather all requirements relating to multiple subscriptions into one section and to improve the clarity.
SMG#24	967/97	006	Section 8.1	3.2.1	To improve the accuracy of text on numbering principles and minor editorial change to section 8.1
SMG#27	98-0551	007	Section 4.6 and misc.	3.3.0	Removal of commercial role model from the specification in order to improve clarity
SMG#27	*98-0552 (Not Approved)*	*008*	*New Section 18 (Not Applied)*	*3.3.0*	*To include requirements for network selection in service principles: NOT APPROVED > NOT APPLIED*
Pre- SMG#28	*(SMG1 Tdoc 98-0893)* 99-040	008 r4 Rejected	New Section 18 Applied	[Draft 3.4.0]	Added Network Selection section - Agreed by correspondence - Jan 13, 1999 - Prepared with CRs applied with revision marks
SMG#27	98-0553	009	Section 4.3	3.3.0	To remove unnecessary reference to IN and B-ISDN
SMG#27	98-0682	010	Section 11	3.3.0	To improve the clarity of service requirements for multiple user profiles
Pre-SMG#28	*(SMG1 Tdoc 98-0869)* 99-040	011	Sections 1, 2, 3, 4, 9, 10, 12, 17	Draft 3.4.0	Clean up for UMTS phase 1 Agreed at SMG1 Rome
Pre-SMG#28	*(SMG1 Tdoc 98-852)* 99-040	012	Sections 3,8,9,11,14,15	Draft 3.4.0	Changes in IC card and terminal service requirements Agreed at SMG1 Rome
Pre-SMG#28	*(SMG1 Tdoc 98-0894)* 99-040	013r1	Section 3.2 & 4.3	Draft 3.4.0	Clarification of general requirements for efficient use of radio resources Agreed by correspondence - Jan 13, 1999 - Prepared with CRs applied with revision marks
Note				Draft 3.4.0	SMG1 agreed only
pre-SMG#28	99-040	015 Rejected	17	Draft 3.4.0	According to the outcome of the SMG 1 ad-hoc meeting on handover issues it is proposed that inter-operator handover is not required for UMTS phase 1.(rejected by smg#28)

Change history					
SMG No.	**TDoc. No.**	**CR. No.**	**Section affected**	**New version**	**Subject/Comments**
SMG#28	99-305	008r5	Revised Section 18	3.4.0	Network Selection presented at SMG#28 in 2201_008r4 was further revised and Approved at SMG#28.
Note				3.4.0	Removal of Section 12 on UPT with CR 011 causes a skip section from Section 11 to 13.

Change history											
TSG SA #	**SA Doc.**	**SA1 Doc**	**Spec**	**CR**	**Rev**	**Rel**	**Cat**	**Subject/Comment**	**Old**	**New**	**WI**
SP-03	SP-99104	S1-99202	22.101	A016		R99	B	Control of supplementary services (GSM 02.04), may use MMI procedures specified in GSM 02.30 and existing GSM MMI related MS features (GSM 02.07) may also be used.	3.4.0		
Post-SA#3			22.101			R99		Updated Logo, …	3.5.0	3.5.1	
SP-04	SP-99229	S1-99387	22.101	021		R99	B	MultiNumbering: It will be possible for multiple MSISDNs to be associated with a single subscription.	3.5.0	3.6.0	
SP-04	SP-99226	S1-99395	22.101	020	7	R99	B	Emergency: To route the call to the appropriate emergency service if more than one emergency number is supported in a country.	3.5.0	3.6.0	
SP-05	SP-99439	S1-99737	22.101	025		R99	B	Support of SAT by USIM	3.6.0	3.7.0	
SP-05	SP-99439	S1-99816	22.101	024		R99	B	Clarification on the usage on 2G SIM and 3G USIM	3.6.0	3.7.0	
SP-05	SP-99435	S1-99851	22.101	022		R99	C	Clarification of Emergency Call requirements	3.6.0	3.7.0	
SP-06	SP-99524	S1-991031	22.101	029		R99	B	Emergency Call	3.7.0	3.8.0	
SP-06	SP-99527	S1-991038	22.101	028		R99	C	FDN	3.7.0	3.8.0	
SP-06	SP-99519	S1-991026	22.101	026		R99	D	Mainly editorial update for GSM/3GPP use.	3.7.0	3.8.0	
SP-07	SP-000060	S1-000112	22.101	030		R99	A	Support of encryption in GPRS mobile stations	3.8.0	3.9.0	
SP-07	SP-000070	S1-000137	22.101	031		R99	F	Fixed Dialing Number (FDN)	3.8.0	3.9.0	
SP-08	SP-000210	S1-000271	22.101	033		R99	D	Network selection procedures removed from section 16, reference to 22.011 added	3.9.0	3.10.0	
SP-08	SP-000200	S1-000350	22.101	035		R99	B	Emergency Calls and numbers used	3.9.0	3.10.0	
SP-08	SP-000201	S1-000362	22.101	038		R99	F	CS multimedia support	3.9.0	3.10.0	
SP-08	SP-000202	S1-000326	22.101	039		R99	F	Clarification for USIM Application selection	3.9.0	3.10.0	
SP-08	SP-000210	S1-000270	22.101	034		R00	D	Network selection procedures removed from section 16, reference to 22.011 added	3.9.0	4.0.0	
SP-08	SP-000200	S1-000351	22.101	036		R00	B	Emergency Calls and numbers used	3.9.0	4.0.0	
SP-08	SP-000213	S1-000352	22.101	037		R00	B	Emergency Call enhancements	3.9.0	4.0.0	
SP-09	SP-000383	S1-000603	22.101	040		R4	B	Multimedia messaging	4.0.0	4.1.0	
SP-09	SP-000383	S1-000605	22.101	041		R4	C	Service Management requirements	4.0.0	4.1.0	
SP-09	SP-000430	S1-000700	22.101	042	1	R4	F	General corrections and clarifications to 22.101 for Release 2000	4.0.0	4.1.0	
SP-09	SP-000383	S1-000598	22.101	046		R4	D	Editorial changes to 22.101 for Release 2000	4.0.0	4.1.0	
SP-09	SP-000430	S1-000698	22.101	047	1	R4	C	Numbering Principles	4.0.0	4.1.0	
SP-09	SP-000383	S1-000620	22.101	048		R4	C	Service evolution	4.0.0	4.1.0	
SP-09	SP-000391	S1-000573	22.101	049		R4	D	Emergency Call	4.0.0	4.1.0	
SP-09	SP-000405	S1-000649	22.101	050		R4	B	Text Conversation	4.0.0	4.1.0	
SP-09	SP-000383	S1-000625	22.101	043		R5	F	Classification of services	4.0.0	5.0.0	
SP-09	SP-000383	S1-000622	22.101	044		R5	B	IP multimedia services	4.0.0	5.0.0	

SP-09	SP-000383	S1-000621	22.101	045		R5	B	IP multimedia session for Emergency call	4.0.0	5.0.0	
SP-09	SP-000430	S1-000699	22.101	051		R5	C	IM Number portability	4.0.0	5.0.0	
SP-09	SP-000430	S1-000701	22.101	052		R5	F	Introduction of IM CN Subsystem	4.0.0	5.0.0	
SP-09	SP-000383	S1-000704	22.101	053		R5	F	Subscription	4.0.0	5.0.0	
SP-09	SP-000383	S1-000705	22.101	054		R5	F	Roaming	4.0.0	5.0.0	
SP-10	SP-000533	S1-000799	22.101	059		Rel-5	A	Deleting Encrypted USIM-ME interface	5.0.0	5.1.0	TEI4
SP-11	SP-010053	S1-010072	22.101	063		Rel-5	A	Handling of interactions between applications requiring the access to UE resources	5.1.0	5.2.0	Service Clean up R99
SP-11	SP-010054	S1-010208	22.101	065		Rel-5	A	PLMN name indication	5.1.0	5.2.0	TEI4
SP-11	SP-010055	S1-010179	22.101	067		Rel-5	A	CR to 22.101 on Introduction of CPHS features	5.1.0	5.2.0	UICC1-CPHS
SP-11	SP-010056	S1-010210	22.101	069		Rel-5	A	Display of service provider name in the UE	5.1.0	5.2.0	TEI4
SP-11	SP-010057	S1-010250	22.101	070		Rel-5	C	CR to 22.101 on Clarifications on IMS emergency call support	5.1.0	5.2.0	EMC1-PS
SP-12	SP-010262	S1-010505	22.101	072		Rel-5	A	Replacement of references to 23.121 for R4 onwards	5.2.0	5.3.0	TEI4
SP-12	SP-010258	S1-010574	22.101	073		Rel-5	C	Subscription and Provisioning	5.2.0	5.3.0	TEI5
SP-12	SP-010255	S1-010577	22.101	075		Rel-5	A	Addition of a Streaming paragraph	5.2.0	5.3.0	PSTREAM
SP-12	SP-010263	S1-010351	22.101	077		Rel-5	A	CS Multimedia fallback to speech	5.2.0	5.3.0	TEI4
SP-12	SP-010253	S1-010595	22.101	080		Rel-5	A	Clarification of PLMN Name Indication and Service Provider Name Indication feature.	5.2.0	5.3.0	SPANME
SP-13	SP-010441	S1-010832	22.101	084		Rel-5	A	Addition of a statement on parameter storage on the SIM/USIM.	5.3.0	5.4.0	TEI4
SP-13	SP-010436	S1-010889	22.101	086	1	Rel-5	F	Definition of Home Environment	5.3.0	5.4.0	VHE1
SP-15	SP-020052	S1-020609	22.101	087		Rel-5	B	CR 22.101 Rel.5 B Service change and fallback for UDI/RDI multimedia calls	5.4.0	5.5.0	SCUDIF
SP-15	SP-020049	S1-020510	22.101	088		Rel-5	C	CR to 22.101on IMS access	5.4.0	5.5.0	43000
SP-15	SP-020051	S1-020613	22.101	089		Rel-5	C	CR to 22.101 on on USIM support in Rel-5 GSM only terminals	5.4.0	5.5.0	UICC1
SP-15	SP-020050	S1-020658	22.101	090		Rel-5	C	CR to 22.101 on Access to IMS services using ISIM Note: special dispensation was given by SA #15 to allow some leeway on the section numbering.	5.4.0	5.5.0	TEI/ISIM
SP-15	SP-020045	S1-020457	22.101	092	-	Rel-5	A	Editorial CR to correct terms and references	5.4.0	5.5.0	CORRECT
SP-15	SP-020126		22.101	093		Rel-5	F	Correction of references to obsolete SIP RFC 2543 IETF specification	5.4.0	5.5.0	IMS-CCR
SP-16	SP-020381		22.101	095	1	Rel-5	F	CR to 22.101 v5.5.0 on REL5 clean up	5.5.0	5.6.0	IMS
SP-16	SP-020255	S1-020848	22.101	096		Rel-6	D	CR to 22.101 v5.5.0 on Editorial for REL6	5.5.0	6.0.0	IMS
SP-17	SP-020557	S1-021849	22.101	103		Rel-6	F	Clarification of SIM support in Rel-6	6.0.0	6.1.0	TEI4
SP-17	SP-020557	S1-021755	22.101	104		Rel-6	B	CR to 22.101 Removal of implementation details for directory number in SMS and other services	6.0.0	6.1.0	TEI6
SP-17	SP-020557	S1-021775	22.101	105		Rel-6	F	CR to 22.101 Rel-6 Clean up of IMS Rel 6 to re-instate requirements	6.0.0	6.1.0	IMS
SP-18	SP-020658	S1-022064	22.101	107		Rel-6	B	CR to 22.101 on IMS number portability rev of 1909	6.1.0	6.2.0	IMS
SP-18	SP-020658	S1-022119	22.101	108		Rel-6	B	CR to 22.101 Rel 6 on Emergency calls	6.1.0	6.2.0	EMC1
SP-18	SP-020666	S1-022263	22.101	109		Rel-6	B	CR to 22.101 on WLAN interworking	6.1.0	6.2.0	WLAN
SP-18	SP-020651	S1-022340	22.101	113		Rel-6	A	CR to 22.101 on Support of SIM and USIM in REL-6	6.1.0	6.2.0	TEI5

SP-19	SP-030022	S1-030215	22.101	114	-	Rel-6	B	Simultaneous connection to 3GPP systems and I-WLANs	6.2.0	6.3.0	WLAN-CR
SP-19	SP-030035	S1-030269	22.101	115	-	Rel-6	B	Requirements for Network Sharing in Rel-6	6.2.0	6.3.0	NTShar-CR
SP-19	SP-030148	S1-030257	22.101	117	-	Rel-6	A	CR to 22.101 Rel 6 on SIM support	6.2.0	6.3.0	TEI5
SP-20	SP-030351		22.101	125	1	Rel-6	A	Alignment of Subscriber Identification requirements to current implementation	6.3.0	6.4.0	TEI5
SP-21	SP-030457	S1-030911	22.101	128	-	Rel-6	A	Clarification on USIM-based access to IMS	6.4.0	6.5.0	IMS
SP-21	SP-030492	S1-031049	22.101	132		Rel-6	C	Cleanup and modifications on identification of emergency numbers in 22.101 Rel-6	6.4.0	6.5.0	EMC1
SP-21	SP-030534	S1-031061	22.101	134	1	Rel-6	A	Support of Release 4 SIM in Release 6	6.4.0	6.5.0	TEI5
SP-22	SP-030700	S1-031339	22.101	135	-	Rel-6	B	Automatic Device Detection	6.5.0	6.6.0	TEI
SP-22	SP-030700	S1-031342	22.101	136	-	Rel-6	C	Correction of Core Network emergency call requirements	6.5.0	6.6.0	EMC1
SP-22	SP-030687	S1-031344	22.101	137	-	Rel-6	C	Clarification of emergency call requirements	6.5.0	6.6.0	EMC1
SP-22	SP-030790	-	22.101	141	-	Rel-6	A	Removal of unnecessary numbers from the ME default emergency number list (Rel-6)	6.5.0	6.6.0	EMC1
SP-23	SP-040084	S1-040198	22.101	145	-	Rel-6	A	Alignment to TS 31.102 on FDN/BDN unsupported terminal procedure.	6.6.0	6.7.0	TEI
SP-23	SP-040091	S1-040215	22.101	146	-	Rel-6	B	Improvements to Circuit Switched Video and Voice Service procedures	6.6.0	6.7.0	CS-VVS
SP-23	SP-040101	S1-040258	22.101	150	-	Rel-6	D	Extraction of redundant WLAN information – now in WLAN TS22.234	6.6.0	6.7.0	WLAN
SP-23	SP-040101	S1-040262	22.101	151	-	Rel-6	D	Extraction of redundant WLAN related simultaneous connection information [now in WLAN TS22.234]	6.6.0	6.7.0	WLAN
SP-24	SP-040288	S1-040427	22.101	152	-	Rel-6	F	Correction of UICC related text.	6.7.0	6.8.0	TEI
SP-24	SP-040292	S1-040537	22.101	154	-	Rel-6	F	Editorial Correction of R5 reference	6.7.0	6.8.0	IMS2
SP-24	SP-040301	S1-040513	22.101	153	-	Rel-7	B	Termination of location privacy override for emergency calls	6.7.0	7.0.0	LCS2; EMC1
SP-27	SP-050058	S1-050161	22.101	158	-	Rel-7	A	Clean-up in Core Network Operator Name Indication section (22.101)	7.0.0	7.1.0	NTShar
SP-27	SP-050059	S1-050252	22.101	160	-	Rel-7	A	Removal of Reference to TS 22.121	7.0.0	7.1.0	TEI7
SP-28	SP-050216	S1-050519	22.101	163	-	Rel-7	A	Clarification on optionality of Service Provider Name indication	7.1.0	7.2.0	TEI-6
SP-28	SP-050225	S1-050581	22.101	164	-	Rel-7	B	Voice call continuity requirements	7.1.0	7.2.0	VCC
SP-29	SP-050522	S1-050793	22.101	0165	-	Rel-7	F	Correction of emergency number example	7.2.0	7.3.0	TEI7
SP-29	SP-050522	S1-050873	22.101	0166	-	Rel-7	F	Requirements on the type of emergency	7.2.0	7.3.0	TEI7
SP-29	SP-050522	S1-050796	22.101	0167	-	Rel-7	C	Modification to chapter 4.2 on service capabilities	7.2.0	7.3.0	TEI7
SP-29	SP-050518	S1-050896	22.101	0168	-	Rel-7	F	Refinement of general description of VCC	7.2.0	7.3.0	VCC
SP-29	SP-050518	S1-050897	22.101	0169	-	Rel-7	C	Charging requirements for voice call continuity	7.2.0	7.3.0	VCC
SP-29	SP-050518	S1-050936	22.101	0171	1	Rel-7	C	Clarification of VCC Triggers	7.2.0	7.3.0	VCC
SP-29	SP-050522	S1-050906	22.101	0172	-	Rel-7	F	Clarification on the identification of emergency numbers	7.2.0	7.3.0	EMC1
SP-29	SP-050522	S1-050883	22.101	0173	-	Rel-7	B	Provisioning parameters stored on USIM	7.2.0	7.3.0	TEI7
SP-29	SP-050518	S1-050934	22.101	0174	-	Rel-7	C	Supplementary Service Requirements for VCC	7.2.0	7.3.0	VCC
SP-30	SP-050750	S1-051145	22.101	0176	-	Rel-7	F	Correcting reference to OMA DM and Client provisioning specifications	7.3.0	7.4.0	TEI7
SP-30	SP-050746	S1-051209	22.101	0177	-	Rel-7	F	Clarification of supplementary servces requirements for VCC	7.3.0	7.4.0	VCC

SP-30	SP-050746	S1-051220	22.101	0178	-	Rel-7	F	Clarification of VCC Triggers	7.3.0	7.4.0	VCC
SP-30	SP-050737	S1-051236	22.101	0179	-	Rel-7	A	Introduction of Cell capability indicator	7.3.0	7.4.0	TEI6
SP-30	SP-050805	-	22.101	0180	1	Rel-7	B	Emergency sip uri to be stored in MT	7.3.0	7.4.0	EMC1
SP-31	SP-060207	-	22.101	0182	1	Rel-8	F	Inclusion of AIPN within the 3GPP service principles specification	7.4.0	8.0.0	AIPN
SP-32	SP-060448	-	22.101	0186	1	Rel-8	F	Corrections to align the Rel-8 and (latest) Rel-7 versions of TS 22.101	8.0.0	8.1.0	TEI8
SP-32	SP-060316	S1-060561	22.101	0187	-	Rel-8	B	VCC Additional flexibility	8.0.0	8.1.0	VCC
SP-32	SP-060315	S1-060621	22.101	0191	-	Rel-8	A	Clarification of IMS voice service	8.0.0	8.1.0	VCC
SP-32	SP-060435	-	22.101	0192	1	Rel-8	B	QoS Parameters Provisioning	8.0.0	8.1.0	TEI8
SP-32	SP-060449	-	22.101	0193	2	Rel-8	B	High Speed Interface between the Terminal and the UICC	8.0.0	8.1.0	TEI8
SP-32	SP-060320	S1-060626	22.101	0194	-	Rel-8	F	Clarification on handling of emergency number	8.0.0	8.1.0	TEI-8
SP-33	SP-060470	S1-060939	22.101	0197	-	Rel-8	A	Clarification on Domain Selection for MO and MT Operations	8.1.0	8.2.0	TEI8
SP-33	SP-060468	S1-060950	22.101	0199	-	Rel-8	A	Emergency calls and ISIM	8.1.0	8.2.0	EMC1
SP-33	SP-060472	S1-060964	22.101	0200	-	Rel-8	B	Requirements for the determination of cell capability usage - when requested by CN	8.1.0	8.2.0	TEI8
SP-34	SP-060777	S1-061321	22.101	0195	3	Rel-8	C	Requirements for the addition of a data component to TS12 emergency calls and eCall MSD (data) transfer requirements	8.2.0	8.3.0	TEMCD
SP-34	SP-060773	S1-061406	22.101	0201	-	Rel-8	B	Serving Environment / Mobile Virtual Network Identification	8.2.0	8.3.0	TEI8
SP-34	SP-060765	S1-061347	22.101	0204	-	Rel-8	A	Removal of IMS emergency call identifier	8.2.0	8.3.0	EMC1
SP-35	SP-070130	S1-070190	22.101	0205	1	Rel-8	D	Addition of Evolved 3GPP System description and corrections to references	8.3.0	8.4.0	SAE-R
SP-35	SP-070134	S1-070299	22.101	0207	1	Rel-8	B	Registration in Densely-populated area	8.3.0	8.4.0	RED
SP-36	SP-070355	S1-070673	22.101	0211	1	Rel-8	C	Graphic format of PLMN name	8.4.0	8.5.0	TEI8
SP-36	SP-070354	S1-070802	22.101	0214	1	Rel-8	A	Alignment of SPDI definition between TS22.101 and TS31.102	8.4.0	8.5.0	TEI5
SP-36	SP-070472	S1-070790	22.101	0215	2	Rel-8	F	Correction of the backward compatibility requirement for the HSP interface	8.4.0	8.5.0	TEI-8
SP-37	SP-070661	S1-071314	22.101	218	2	Rel-8	B	ICS Requirements – General	8.5.0	8.6.0	ICS-RA
SP-37	SP-070661	S1-071315	22.101	221	2	Rel-8	B	ICS Requirements - Service Consistency	8.5.0	8.6.0	ICS-RA
SP-37	SP-070661	S1-071316	22.101	222	2	Rel-8	B	ICS Requirements - Service Continuity	8.5.0	8.6.0	ICS-RA
SP-37	SP-070661	S1-071317	22.101	225	2	Rel-8	B	ICS Requirements - IMS Services	8.5.0	8.6.0	ICS-RA
SP-37	SP-070662	S1-070995	22.101	216	1	Rel-8	C	Clarification on graphic format of PLMN name	8.5.0	8.6.0	TEI8
SP-37	SP-070676	-	22.101	217	4	Rel-8	B	Requirements for eCall	8.5.0	8.6.0	EData
SP-38	SP-070850	S1-071712	22.101	0232	1	Rel-8	B	ICS - Clarification of Service Continuity	8.6.0	8.7.0	ICSRA
SP-38	SP-070850	S1-071713	22.101	0233	1	Rel-8	B	ICS - Service Continuity Requirements	8.6.0	8.7.0	ICSRA
SP-38	SP-070850	S1-071714	22.101	0234	1	Rel-8	B	ICS - Roaming Support	8.6.0	8.7.0	ICSRA
SP-38	SP-070850	S1-071715	22.101	0235	1	Rel-8	B	Clarify UE requirements for ICS	8.6.0	8.7.0	ICS-RA
SP-38	SP-070861	S1-071932	22.101	0229	2	Rel-8	B	Requirements for Emergency Call-Back	8.6.0	8.7.0	TEI8
SP-38	SP-070862	S1-071927	22.101	0241	2	Rel-8	B	CS IP Interconnection requirement	8.6.0	8.7.0	IPINTERC
SP-39	SP-080036	S1-080309	22.101	0237	6	Rel-8	B	Requirements for "In Case of Emergency" (ICE) information	8.7.0	8.8.0	ICE
SP-39	SP-080042	S1-080296	22.101	0250	1	Rel-8	F	Update the related contents about "Evolved Packet System"	8.7.0	8.8.0	AIPN-SAE

SP-39	SP-080041	S1-080297	22.101	0251	1	Rel-8	F	CS multimedia calls	8.7.0	8.8.0	TEI8
SP-39	SP-080041	S1-080298	22.101	0252	1	Rel-8	D	correction of wrong referencing	8.7.0	8.8.0	TEI8
SP-39	SP-080032	S1-080314	22.101	0253	2	Rel-8	B	Requirements for transfer for data during emergency calls	8.7.0	8.8.0	EData
SP-39	SP-080032	S1-080266	22.101	0254	1	Rel-8	B	Requirements for the transfer of eCall Minimum Set of Data	8.7.0	8.8.0	EData
SP-39	SP-080043	S1-080337	22.101	0255	2	Rel-8	B	RAT indicator	8.7.0	8.8.0	TEI8
SP-39	SP-080039	S1-080287	22.101	0256	1	Rel-8	B	ICS requirement for roaming	8.7.0	8.8.0	ICSRA
SP-39	SP-080040	S1-080237	22.101	0262	1	Rel-8	B	Additional requirements for CS IP interconnection	8.7.0	8.8.0	IPINTERC
SP-39	SP-080043	S1-080299	22.101	0263	1	Rel-8	B	Addition of Unique Device Identifier to Calling Line Identification	8.7.0	8.8.0	TEI8
SP-39	SP-080040	S1-080327	22.101	0265	1	Rel-8	B	Clarification of requirements for CS IP interconnection	8.7.0	8.8.0	IPINTERC
SP-40	SP-080308	S1-080746	22.101	0267	3	Rel-8	F	Emergency Call Interaction	8.8.0	8.9.0	EMC1
SP-40	SP-080299	S1-080581	22.101	0269	1	Rel-8	C	eCall call back requirements clarification	8.8.0	8.9.0	EDATA
SP-40	SP-080298	S1-080735	22.101	0270	3	Rel-8	F	Common IMS Proposals - Requirements for Emergency Calls	8.8.0	8.9.0	CIMS_3GPP2
SP-40	SP-080306	S1-080715	22.101	0271	1	Rel-8	F	Correction of reference for PLMN architecture	8.8.0	8.9.0	AIPN-SAE
SP-40	SP-080299	S1-080714	22.101	0272	1	Rel-8	F	Corrections to general requirements for emergency calls	8.8.0	8.9.0	EData
SP-40	SP-080308	S1-080560	22.101	0274	-	Rel-8	F	Clarification of IMS Emergency Calls	8.8.0	8.9.0	EMC1
SP-40	SP-080299	S1-080743	22.101	0275	1	Rel-8	C	Reconfigurable eCall only terminal access to test and reconfiguration services	8.8.0	8.9.0	EDATA
SP-40	SP-080302	S1-080705	22.101	0276	-	Rel-8	F	Correction of ICE requirements	8.8.0	8.9.0	ICE
SP-40	SP-080311	S1-080774	22.101	0268	3	Rel-9	B	Requirements for Service Alignment and Migration	8.8.0	9.0.0	ETWS-S1
SP-41	SP-080498	S1-082192	22.101	0282	1	Rel-9	F	Clarification on IMS services supported by ICS	9.0.0	9.1.0	ICSRA
SP-41	SP-080643	-	22.101	0280	2	Rel-9	F	RED optimization of location update	9.0.0	9.1.0	RED
SP-42	SP-080884		22.101	0290	6	Rel-9	A	ICS Service Continuity	9.1.0	9.2.0	CIMS8
SP-42	SP-080779	S1-084415	22.101	281	2	Rel-9	B	User Data Convergence Requirements	9.1.0	9.2.0	UDC
SP-42	SP-080781	S1-084177	22.101	0283	2	Rel-9	F	VCC for emergency calls	9.1.0	9.2.0	IMS_EMER_GPRS_EPS
SP-42	SP-080784	S1-084424	22.101	0290	3	Rel-9	C	ICS Service Continuity	9.1.0	9.2.0	TEI9
SP-42	SP-080783	S1-084404	22.101	0299	3	Rel-9	B	UICC applications access to IMS	9.1.0	9.2.0	TEI9
SP-42	SP-080781	S1-084096	22.101	300	-	Rel-9	F	Make IMS emergency calls available in all access systems	9.1.0	9.2.0	TEI9
SP-43	SP-090079	S1-090174	22.101	0307	1	Rel-9	A	Clarification and enhancement of ciphering indicator feature	9.2.0	9.3.0	TEI9
SP-44	SP-090377	S1-091407	22.101	0312	4	Rel-9	F	Domain selection for emergency call	9.3.0	9.4.0	IMS_EMER_GPRS_EPS
SP-44	SP-090369	S1-091393	22.101	0313	-	Rel-9	A	eCall - PSAP acknowledgement to the IVS	9.3.0	9.4.0	Edata
SP-45	SP-090475	S1-093290	22.101	0315	1	Rel-9	A	Minimise delay to emergency voice calls	9.4.0	9.5.0	EDATA
SP-45	SP-090480	S1-093282	22.101	321	1	Rel-9	F	Domain selection for emergency call	9.4.0	9.5.0	IMS_EMER_GPRS_EPS
SP-45	SP-090483	S1-093286	22.101	0316	1	Rel-10	F	Add reference to MSD specification	9.4.0	10.0.0	EDATA
SP-45	SP-090484	S1-093341	22.101	0320	4	Rel-10	B	Selected IP Traffic Offload	9.4.0	10.0.0	LIPA_SIPTO
SP-45	SP-090483	S1-093261	22.101	322	1	Rel-10	F	Clarification of emergency call back requirements	9.4.0	10.0.0	TEI10
SP-46	SP-090887	S1-094259	22.101	0336	1	Rel-10	A	Providing eCall indication to the PSAP	10.0.0	10.1.0	EDATA

SP-46	SP-090836	S1-094262	22.101	0338	-	Rel-10	A	Remove requirement for operator determined eCall call-back duration.	10.0.0	10.1.0	EDATA
SP-46	SP-090842	S1-094272	22.101	0330	1	Rel-10	A	User data repository shall be possible to be shared among different PLMNs that have trusted relationships	10.0.0	10.1.0	UDC
SP-46	SP-090844	S1-094499	22.101	0327	2	Rel-10	A	Use of GTT-IP for IMS emergency calls (mirror)	10.0.0	10.1.0	TEI9
SP-46	SP-090846	S1-094496	22.101	0328	1	Rel-10	B	ICS Enhancements	10.0.0	10.1.0	TEI10
SP-46	SP-090849	S1-094300	22.101	0325	3	Rel-10	B	Roaming and Mobility aspects for Selected IP Traffic Offload	10.0.0	10.1.0	LIPA_SIPTO
SP-47	SP-100187	S1-100321	22.101	0339	2	Rel-10	B	SIPTO requirements common for macro network and H(e)NB subsystems	10.1.0	10.2.0	LIPA_SIPTO
SP-47	SP-100188	S1-100445	22.101	0332	6	Rel-10	C	Clarification to Emergency call selection prioritization	10.1.0	10.2.0	TEI10
SP-47	SP-100188	S1-100442	22.101	0340	1	Rel-10	C	Clarification on ICS wrt data (CS) and fax	10.1.0	10.2.0	TEI10
SP-47	SP-100191	S1-100431	22.101	0343	2	Rel-10	B	IMS emergency call enhancements	10.1.0	10.2.0	IESE
SP-48	SP-100431	S1-101060r	22.101	0347	6	Rel-10	B	Addition of Requirement for UDC Data Model	10.2.0	10.3.0	TEI10
SP-48	SP-100401	S1-101067	22.101	0350	-	Rel-10	F	Correction of reference	10.2.0	10.3.0	IESE
SP-49	SP-100585	S1-102403	22.101	0353	2	Rel-11	B	SIPTO mobility/continuity	10.3.0	11.0.0	SIPTO_SC
SP-51	SP-110168	S1-110088	22.101	0360	-	Rel-11	F	Clarification on Service Continuity for SIPTO	11.0.0	11.1.0	SIPTO_SC
SP-52	SP-110373	S1-111410	22.101	0367	3	Rel-11	B	IMS emergency call requirements for additional media types	11.1.0	11.2.0	NOVES
SP-52	SP-110376	S1-111209	22.101	0369	-	Rel-11	D	Removing text discrepancy between Rel-10 and Rel-11 for UDC Data Model	11.1.0	11.2.0	TEI11
SP-52	SP-110418	-	22.101	0372	2	Rel-11	A	IMS emergency calls	11.1.0	11.2.0	IMS_EMER_GPRS_EPS
								Correction of typo on cover page	11.2.0	11.2.1	
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SP-53	SP-110576	S1-112101	22.101	0379	-	Rel-11	B	IMS emergency call support with other media handover considerations	11.2.0	11.3.0	NOVES
SP-53	SP-110576	S1-112344	22.101	0380	1	Rel-11	B	Notification of IMS emergency call support with other media	11.2.0	11.3.0	NOVES
SP-53	SP-110576	S1-112350	22.101	0385	3	Rel-11	B	Domain Selection for IMS emergency calls with additional media types	11.2.1	11.3.0	NOVES
SP-53	SP-110579	S1-112385	22.101	0374	1	Rel-11	F	SIPTO Service Continuity	11.2.0	11.3.0	SIPTO_SC
SP-53	SP-110579	S1-112393	22.101	0378	2	Rel-11	C	Clarification on collection of traffic and signalling for SIPTO	11.2.0	11.3.0	SIPTO_SC
SP-53	SP-110582	S1-112025	22.101	0376	-	Rel-11	F	Correction of clause numbers and editorial corrections	11.2.1	11.3.0	TEI11
SP-53	SP-110651	S1-112328	22.101	0383	3	Rel-11	F	Moving "Network provided destination for uplink data" requirement from 22.368 into 22.101	11.2.1	11.3.0	SIMTC
SP-53	SP-110651	S1-112329	22.101	0384	4	Rel-11	F	Moving PS-Only requirements from 22.368 into 22.101	11.2.1	11.3.0	SIMTC
SP-54	SP-110811	S1-113245	22.101	0387	1	Rel-11	F	IMS Multimedia Emergency Session support for UEs in Limited Service Mode	11.3.0	11.4.0	NOVES
SP-54	SP-110811	S1-113249	22.101	0388	2	Rel-11	F	Correction and removal of inconsistencies to IMS Emergency call when using other media	11.3.0	11.4.0	NOVES
SP-54	SP-110811	S1-113248	22.101	0390	2	Rel-11	F	Indication of supported media during an IMS multimedia emergency session	11.3.0	11.4.0	NOVES
SP-54	SP-110811	S1-113250	22.101	0391	1	Rel-11	F	Handover of other media to networks that do not support IMS emergency voice calls	11.3.0	11.4.0	NOVES

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SP-54	SP-110811	S1-113094	22.101	0393	-	Rel-11	F	Routing IMS multimedia sessions towards the PSAP	11.3.0	11.4.0	NOVES
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SP-56	SP-120286	S1-121314	22.101	0409	1	Rel-12	A	Emergency calls from non CSG members in CSG cells	12.0.0	12.1.0	EHNB
SP-56	SP-120288	S1-121396	22.101	0412	1	Rel-12	A	Clarification of PS additional numbers requirement	12.0.0	12.1.0	TEI11
SP-56	SP-120288	S1-121392	22.101	0415	-	Rel-12	A	Modification of how ME treats emergency call type(s)	12.0.0	12.1.0	TEI11
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SP-57	SP-120528	S1-122433	22.101	0417	2	Rel-12	B	Requirements for integration of Single Sign-On (SSO) frameworks with 3GPP networks	12.1.0	12.2.0	SSO_int
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SP-59	SP-130112	S1-131226	22.101	0445	3	Rel-12	B	Add Requirement on emergency IM CN subsystem	12.3.0	12.4.0	ESID
SP-61	SP-130413	S1-134180	22.101	0463	1	Rel-13	B	E-UTRAN Sharing Enhancements	12.4.0	13.0.0	RSE
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SP-62	SP-130601	S1-135309	22.101	0465	3	Rel-13	B	Support for Load Balancing in Shared E-UTRAN	13.0.0	13.1.0	RSE
SP-62	SP-130601	S1-135310	22.101	0466	3	Rel-13	B	Additional RAN Sharing Requirements	13.0.0	13.1.0	RSE
SP-62	SP-130601	S1-135294	22.101	0468	2	Rel-13	C	Clarification of Hosting E-UTRAN Operator role	13.0.0	13.1.0	RSE
SP-62	SP-130601	S1-135313	22.101	0469	3	Rel-13	C	Adding explanatory text to requirements on E-UTRAN Sharing	13.0.0	13.1.0	RSE
SP-62	SP-130601	S1-135312	22.101	0470	3	Rel-13	B	Enabling Dynamic capacity negotiation	13.0.0	13.1.0	RSE
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SP-65	SP-140501	S1-143604	22.101	0473	5	Rel-13	B	Exposure of network service and capabilities to 3rd party	13.2.0	13.3.0	SEES
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SP-65	SP-140510	S1-143556	22.101	0478	3	Rel-13	B	Requirements for Flexible Mobile Steering Service	13.2.0	13.3.0	FMSS
SP-66	SP-140755	S1-144611	22.101	481	2	Rel-13	B	GERAN & UTRAN Sharing, change to the title of Section 28 and update to clause 28.1	13.3.0	13.4.0	GUSH
SP-66	SP-140755	S1-144617	22.101	482	2	Rel-13	B	GERAN & UTRAN Sharing, new clauses 28.3 and 28.3.1 (Allocation of Shared GERAN & UTRAN requirements).	13.3.0	13.4.0	GUSH
SP-66	SP-140755	S1-144618	22.101	483	2	Rel-13	B	GERAN & UTRAN Sharing, new clause 28.3.2 (OA&M access to the shared RAN).	13.3.0	13.4.0	GUSH
SP-66	SP-140755	S1-144533	22.101	484	1	Rel-13	B	GERAN & UTRAN Sharing, new clause 28.3.3 (Generation and retrieval of usage and accounting information).	13.3.0	13.4.0	GUSH
SP-66	SP-140755	S1-144534	22.101	485	1	Rel-13	B	GERAN & UTRAN Sharing, new clause 28.3.4 (MDT Collection)	13.3.0	13.4.0	GUSH
SP-66	SP-140755	S1-144640	22.101	486	3	Rel-13	B	GERAN & UTRAN Sharing, new clause 28.3.5 (PWS Support)	13.3.0	13.4.0	GUSH

SP-66	SP-140755	S1-144619	22.101	487	2	Rel-13	B	GERAN & UTRAN Sharing, new clause 28.3.6 (Support for load balancing).	13.3.0	13.4.0	GUSH
SP-66	SP-140755	S1-144636	22.101	488	3	Rel-13	B	GERAN & UTRAN Sharing, new clause 28.3.7 (Dynamic capacity negotiation).	13.3.0	13.4.0	GUSH
SP-66	SP-140781	S1-144528	22.101	493	-	Rel-13	A	eCall mode of operation	13.3.0	13.4.0	EDATA
SP-66	SP-140751	S1-144556	22.101	494	-	Rel-13	F	Align the MTC monitoring requirements with Service Exposure requirements	13.3.0	13.4.0	TEI13
SP-67	SP-150039	S1-150214	22.101	495	2	Rel-13	F	Update of Definitions	13.4.0	13.5.0	GUSH
SP-69	SP-150536	S1-152751	22.101	502	2	Rel-13	F	Support of Emergency sessions over WLAN	13.5.0	13.6.0	TEI-13
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SP-71	SP-160094	S1-160471	22.101	0514		Rel-13	F	Clarification on Support of Emergency sessions over WLAN	13.7.0	13.8.0	TEI13
SP-72	SP-160353	S1-161593	22.101	517	2	Rel-13	B	Addition of Cellular IoT non-IP transport to exposed services and capabilities	13.8.0	13.9.0	SEES
SP-75	SP-170149	S1-171436	22.101	0523	2	Rel-13	F	Alignment of UPCON Service Requirements	13.9.0	13.10.0	UPCON

History

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