Technical Specification Group Services and System Aspects
Meeting #6, Nice, France, 15-17 December 1999

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Source: **TSG SA1**

Title: **Various editorial change requests to SA1 3G Specifications**

Document for: **Approval**

Agenda Item: **5.1.4**

Status	Spec	CR	Rev	Phase	Subject	CAT	Vers	New Vers	TSG Meeting	TSG Doc.No.	Pres
	22.001	001		R99	Mainly an editorial update for GSM/3GPP use	D	3.0.0	3.1.0	S1#06	S1-991076	No
	22.003	001		R99	Mainly an editorial update for GSM/3GPP use	D	3.0.0	3.1.0	S1#06	S1-991025	No
	22.030	006		R99	Mainly an editorial update for GSM/3GPP use.	C	3.1.0	3.2.0	S1#06	S1-991029	No
	22.081	002		R99	Editorial update to TS 22.081 in order to include 3G systems	D	3.0.1	3.1.0	S1#06	S1-99908	No
	22.085	002		R99	Editorial update for GSM/3GPP use.	D	3.0.1	3.1.0	S1#06	S1-99999	No
	22.101	026		R99	Mainly editorial update for GSM/3GPP use.	D	3.7.0	3.8.0	S1#06	S1-991026	No
	22.105	021		R99	Mainly editorial update for GSM/3GPP use.	D	3.6.0	3.7.0	S1#06	S1-991027	No

TSG-SA Working Group 1 meeting #6 **TSG S1 *(99) 1076***

San Diego, 29 Nov – 03 Dec 1999 Agenda: 6.0

3G CHANGE REQUEST

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

22.001 CR 001 Current Version: 3.0.0

3G specification number ↑ *↑ CR number as allocated by 3G support team*

For submision to TSG SA#6 for approval [X] for information [ ] *(only one box should be marked with an X)*

list TSG meeting no. here ↑

Proposed change affects: USIM [X] ME [X] UTRAN [X] Core Network [X]

(at least one should be marked with an X)

Source:	SA WG1/SMG1
Date:	
Subject:	Mainly an editorial update for GSM/3GPP use.
3G Work item:	

Category: *(only one category shall be marked with an X)*

F	Correction	
A	Corresponds to a correction in a 2G specification	
B	Addition of feature	
C	Functional modification of feature	
D	Editorial modification	X

Reason for change: The transfer of GSM specifications for 3GPP requires an editorial update. Text referring to the GSM system needs to be changed to refer to both the GSM and 3G systems. 02.01 is proposed to be transferred to 22.001 with this CR. 02.01 describes only the Circuit services requirements. Requirements covered already by 22.101 and 22.105 has been deleted.

Clauses affected: All clauses

Other specs affected:

Other 3G core specifications	[ ]	→ List of CRs:
Other 2G core specifications	[ ]	→ List of CRs:
MS test specifications	[ ]	→ List of CRs:
BSS test specifications	[ ]	→ List of CRs:
O&M specifications	[ ]	→ List of CRs:

Other comments:

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~~GSM 0~~TS 22.001 V3.0.0~~8.1.0642~~ (1999-0

Technical Specificat

Digital cellular telecommunications system (Phase 2+);
Principles of circuit telecommunication services supported by a ~~GSM~~
Public Land Mobile Network (PLMN);
(TS 22.001~~GSM 02.01~~ version 3~~8~~.0~~1~~.0 Release 1999)

Available SMG only

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Reference

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0 Scope

This TS covers the definition of the circuit telecommunication services supported by a ~~GSM~~ PLMN. The purpose of this TS is to provide a method for the characterization and the description of these telecommunication services.

TS 22.101 describes overall service principles of a PLMN.

0.1 References

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

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.
- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.
- For this Release 1999 document, references to GSM documents are for Release 1999 versions (version 8.x.y).

[1] GSM 01.04: "Digital cellular telecommunications system (Phase 2+); Abbreviations and acronyms"

-[2] ITU-T Recommendation I.221: "Common specific characteristics of services".

[3] ITU-T Recommendation X.200: "Information technology - Open Systems Interconnection - Basic reference model: The basic model".

[4] TS 22.101: "UMTS Service Principles".

[5] TS 22.002: "Bearer services supported by a PLMN".

[6] TS 22.003: "Teleservices supported by a PLMN".

[7~~6~~] TS 22.004: "General on Supplementary Services".

[8~~7~~] TS 27.001: "-General on Terminal Adaptation Functions (TAF) for Mobile Stations (MS)".

[9~~8~~] TS 22.030: "Man-Machine Interface (MMI) of the ~~Mobile Station~~User equipment (MS)"

[10~~9~~] TS 22.081: "Line Identification Supplementary Servicess; Stage 1"

~~[10] TS 22.060: "General Packet Radio Service"~~[11] TS 22.135: "Multicall; Stage 1"

[12] TR 21.905: "Vocabulary for 3GPP Specifications"

[13] GSM 04.08: "Digital cellular telecommunications system (Phase 2+); Mobile radio interface layer 3 specification"

0.2 Abbreviations

Abbreviations used in this TS are listed in GSM 01.04 and TR 21.905 [12].

1 Framework for the description of telecommunication services

~~1.1 General~~

~~Telecommunication services supported by a GSM PLMN are the communication capabilities made available to customers by network operators. A GSM PLMN provides, in cooperation with other networks, a set of network capabilities which are defined by standardized protocols and functions and enable telecommunication services to be offered to customers.~~

~~A service provision by a network operator (e.g. an Administration or an RPOA) to a subscriber of a GSM PLMN may cover the whole or only part of the means required to fully support the service. The operational and commercial features associated with the provision of the service are included in the service concept.~~

~~The service classification and description which follow are independent of different possible arrangements for the ownership and provision to the customer of the means required to support a service.~~

1.1~~2~~ The attribute method of characterization of circuit telecommunication services

This characterization is made by using a set of attributes. A telecommunication service attribute is a specific characteristic of that service whole values distinguish it from other telecommunication services. Particular values are assigned to each attribute when a given telecommunication service is described and defined.
A list of definitions of attributes and values used for bearer services and teleservices is contained in, respectively, annex A and annex B.

2 Description of circuit telecommunication services by the attribute method

2.1 General

Telecommunication services are described by attributes which define service characteristics as they apply at a given reference point where the customer accesses the service. The description of a telecommunication service by the method of attributes is composed of:

- technical attributes as seen by the customer, and;

- other attributes associated with the service provision, e.g. operational and commercial attributes.

~~2.2 Bearer services and teleservices~~

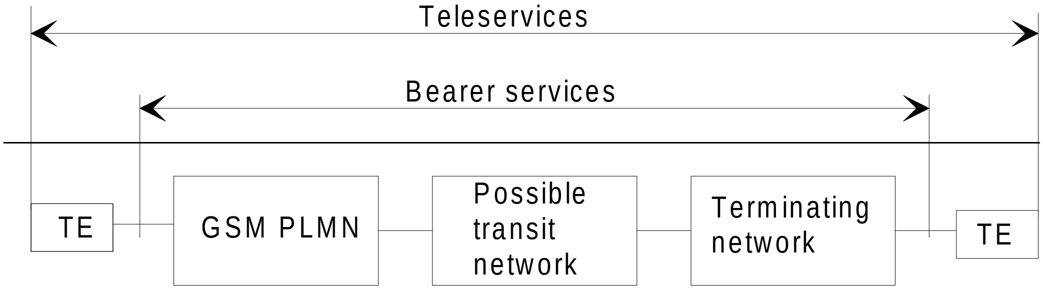
~~Telecommunication services are divided in two broad categories:~~

- ~~bearer services, which are telecommunication services providing the capability of transmission of signals~~

~~between access points;~~

- ~~teleservices, which are telecommunication services providing the complete capability, including terminal equipment functions, for communication between users according to protocols established by agreement between network operators.~~

~~Figure 1 illustrates these definitions.~~



~~TE: Terminal Equipment~~

~~Figure 1: Bearer services and Teleservices supported by a GSM PLMN~~

~~NOTE 1: In the majority of cases, at least two networks of different types are involved in the support of a telecommunication service.~~

~~NOTE 2: Figure 1 does not preclude any routing possibility.~~

~~NOTE 3: In order to limit the complexity of the figure, only one transit network is shown.~~

~~NOTE 4: The terminating network type may include a GSM PLMN, either the originating one or another one.~~

~~2.3 Supplementary services~~

~~A supplementary service modifies or supplements a basic telecommunication service. Consequently, it cannot be offered to a customer as a stand alone service. It must be offered together or in association with a basic telecommunication service. The same supplementary service may be applicable to a number of telecommunication services.~~

~~NOTE: Supplementary services are characterized by the attribute method (see TS 22.004 [6]).~~

2.24 Categorisation of telecommunication services

The concepts introduced in this TS are illustrated in table1.

Table 1: Categorisation of telecommunication services

TELECOMMUNICATION SERVICES			
BEARER SERVICE		TELESERVICE	
Basic Bearer Service	Basic Bearer service + supplementary services	Basic Teleservice	Basic Teleservice + supplementary service

~~3.4 Virtual Home Environment~~

~~The user may be provided with a comprehensive set of services and features which have the "same look and feel" wherever they are used, this concept is called Virtual Home Environment (VHE) TS 22.101 [4]. VHE maybe supported on GSM telecommunication services and service capabilities.~~

~~GSM phase 2+ includes the standardisation of service capabilities in addition to supplementary services. Service capabilities consist of bearers 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. Mechanisms supported are e.g. Customised Application for Mobile network Enhanced Logic (CAMEL), SIM application toolkit (SAT) and Mobile Station Execution Environment (MExE). It is intended that standardised capabilities should provide a defined platform which will enable the support of speech, video, multi-media, messaging, data, other teleservices, user applications and supplementary services.~~

~~The standard shall support access to the same services and features of a user's VHE via the GSM BSS as may be available via the UMTS UTRAN, subject to the performance limitations of the GSM access network.~~

~~4 Capabilities to support a telecommunication service~~

~~4.1 General~~

~~The capabilities to fully support a telecommunication service for a customer accessing a GSM PLMN include:~~

- ~~network capabilities (in the GSM PLMN and in most cases in another network);~~
- ~~terminal capabilities, when required;~~
- ~~other service providing capabilities, when required;~~
- ~~operational and commercial features associated with the service provision.~~

~~4.2 Network capabilities~~

~~Two different levels of GSM PLMN network capabilities are introduced:~~

- ~~low layer capabilities, which relate to bearer services;~~
- ~~high layer capabilities, which together with low layer capabilities relate to teleservices.~~

~~4.3 Terminal capabilities~~

~~Terminal capabilities are also described in terms of low layer and high layer capabilities. In the description of teleservices, the terminal capabilities, both low layer and high layer, are included in the service definition. In the case of bearer service definition, the terminal capabilities are not included but the terminal equipment must conform to the low layer capabilities of the bearer service.~~

~~4.4 Operational capabilities~~

~~The operational capabilities associated with a service offering may include capabilities for maintenance, charging, user control of service features, etc.~~
~~The use of such capabilities may involve terminal-network communication and may therefore be viewed as specific applications.~~

3~~5~~ Characterization of circuit telecommunication services

3~~5~~.1 General

A telecommunication service supported by a ~~GSM~~ PLMN is characterized and described by service attributes.
There are two groups of service attributes applicable to user information flow:

- low layer attributes;

- high layer attributes.

Bearer services are characterized only by low layer attributes. Teleservices are characterized by both low layer attributes and high layer attributes.
The basic characteristics of a telecommunication service are described by the basic service attributes.
The additional characteristics associated with a supplementary service which modify or supplement a basic telecommunication service are described in TS 22.004 [7~~6~~]~~.~~.

3~~5~~.2 Bearer services ~~supported by a GSM PLMN~~

~~Bearer services supported by a GSM PLMN provide the capability for information transfer between a GSM PLMN access point 1 or 2 and an appropriate access point in a terminating network and involve only low layer functions (i.e. relating to layers 1-3 of the OSI Reference Model).~~
~~The customer may choose any set of high layer (at least 4-7) protocols for his communication, but a GSM PLMN will not ensure compatibility at these high layers between customers.~~
Bearer services are characterized by a set of low layer attributes in TS 22.002 [5]. These attributes are classified into four categories:

- information transfer attributes;
- access attributes;
- interworking attributes;
- general attributes, including operational and commercial attributes.

The bearer capability defines the technical features of a bearer service as they appear to the user at the appropriate access point. For the time being, the bearer capability is characterized by information transfer, access and interworking attributes. A bearer capability is associated with every bearer service.

~~The bearer service provides the user with the possibility of gaining access to various forms of communication, covering for example:~~

- ~~information transfer between a user in a GSM PLMN and a user in a terminating network, including the same GSM PLMN, another GSM PLMN and other types of PLMNs;~~

- ~~information transfer between a user in a GSM PLMN and separate resource providing high layer functions.~~

3~~5~~.3 Teleservices ~~supported by a GSM PLMN~~

Circuit ~~T~~teleservices provide the full capacity for communication by means of terminals and network functions and possibly functions provided by dedicated centres.

~~A teleservice supported by a GSM PLMN should use only one (or a small number of) bearer capability recommended by GSM.~~ Circuit~~GSM~~ teleservices are specified in TS ~~GSM 0~~22.003 [6].Teleservices are characterized by a set of low layer attributes, a set of high layer attributes and operational and commercial attributes.

Low layer attributes are those used to characterize the bearer capability ~~(see subclause 5.2)~~. High layer attributes are used in Specification TS 22.003 [6]~~GSM 02.03~~ to describe high layer (i.e. layer 4-7) information transfer related characteristics. They refer to functions and protocols of layers 4-7 in the ITU-T Recommendation X.200 framework which are concerned with the transfer, storage and processing of user messages (provided by a subscriber's terminal, a retrieval centre or a network service centre).

Therefore, not all attributes can be applied directly at the user to terminal interface as they represent two kinds of features, the bearer capability and the terminal features, that are not directly perceived by the user.

~~A teleservice provides the user with the possibility of gaining access to various forms of applications (or teleservice APPLICATIONS) covering for example:~~

- ~~teleservice application involving two terminals providing compatible or identical teleservice attributes at an access point in a GSM PLMN and an access point in a terminating network;~~

- ~~teleservice application involving a terminal at one access point in a GSM PLMN and a system providing high layer functions (e.g. speech storage system, message handling system) located either within the GSM PLMN or in a terminating network.~~

4~~6~~ Provision of telecommunication services

The provision of telecommunication services implies:

- subscription of basic services and possibly subscription to supplementary services;

- registration into a service directory;

- compatibility between terminals;

- interworking capabilities.

The user's subscription to a Basic or Supplementary service is normally verified by the network prior to completion of Call Establishment and/or Supplementary Service operation. This subscription checking shall be performed in accordance with the following sections.

4~~6~~.1 Subscription checking for Basic Services

General

Subscription checking is the function/process to ascertaining whether a subscriber has the authorization to use the particular Basic Service deduced from the call set-up parameters. It is the responsibility of the HPLMN to transfer, to the VPLMN, only the subscription data corresponding to those services a given subscriber is entitled to use in that VPLMN.

For mobile originated calls, subscription checking is performed in the VLR, whilst for mobile terminated calls it is performed in either the HLR or the VLR (determined as described below). The prerequisite for executing the subscription check is a successful deduction of a Basic Service from the Compatibility Information contained in the call set up, i.e. Bearer Capability Information Element and, in some cases, also the Low Layer and High Layer Compatibility Information elements.

For mobile originated calls an ~~MS~~UE shall indicate the requested service by appropriate compatibility information elements according to TS 27.001 [8~~7~~]. This information is mapped to an individual Basic Service code (i.e. the MAP representation) by the MSC in order to be compared with the subscriber data available in the VLR.

An equivalent process is required in the HLR for mobile terminated calls, where the caller's requested service is indicated to the HLR (by the ISDN) by exhaustive compatibility information consisting of ISDN Bearer Capability Information Elements and in some cases - depending on the service requested - also of Low Layer and High layer Compatibility information elements. In case the compatibility information is not exhaustive, e.g. when the call is originated/transited by a PSTN, no ~~GSM~~ Basic Service can be deduced and subscription checking cannot be performed in the "normal" way. Instead, rules for the Single and Multi Numbering Schemes apply.

In the Multi Numbering Scheme the Basic Service can be deduced by information stored in the HLR against the called number and hence an implicit subscription check is performed. In the Single Numbering Scheme, the Basic Service cannot be deduced until the ~~MS~~UE has responded to the set up and therefore the HLR cannot perform subscription check. Instead, the VLR/MSC will perform the subscription check or calls are passed "unfiltered" (as regards subscription check), at the network operators' discretion.

Bearer Services
TS 22.002 [5] lists the Bearer Services, each of them with a specific "BS number". Single services defined independent of the fixed network user rate are called General Bearer Services. These distinct [numbered] services may individually be provided to a subscriber. Whichever the subscription arrangements are, all PLMNs (MSCs, VLRs and HLRs) shall be able to allow - as regards subscription checking - the use of individually subscribed-to Basic Services, within the range of services supported by the PLMN. That is, whenever it is possible to deduce the Basic Service from a call set up, subscription check shall be performed at the granularity of that particular Basic Service or the group to which it belongs.
TeleServices
TS 22.003 [6] ~~GSM 02.03~~ lists the TeleServices, each of them with a specific "TS number". These may be provided to subscribers individually or combined, to the operators' discretion, however TS 12 (E-calls) and 23 (CB) are not subscribable. But, as for Bearer Services, networks shall be able to handle subscription checking at the granularity of individual TeleServices.
Table 2 summarizes the basis on which a successful subscription checking will result. It also describes on which basis Supplementary Service handling for a given call set-up should be performed.

Table 2

Set Up	Subscription Check	SS handling
BS 20	BS 20	BS Group 2x
BS 30	BS 30	BS Group 3x
~~BS 70~~	~~BS 70~~	~~N.A.~~
TS 11	TS 11, TS Group 1x or TS Group All	TS Group 1x
TS 12	N.A.	
TS 21	TS 21, TS Group 2x or TS Group All	TS Group 2x
TS 22	TS 22, TS Group 2x or TS Group All	TS Group 2x
TS 23	N.A.	
TS 61	TS 61, TS Group 6x or TS Group All	TS Group 6x
TS 62	TS 61, 62, Group 6x or TS Group All	TS Group 6x
TS 91	TS 91, TS Group 9x or TS Group All	TS Group 9x
TS 92	TS 92, TS Group 9x or TS Group All	TS Group 9x

Legend:

Set up: The Basic Service which is set up for the call.

Subscription check: Required VLR or HLR data for successful subscription check.

SS handling: Against which VLR or HLR data SS handling should be performed. For example; a call set-up indicating BS61 and Asynchronous mode should be treated for SS purposes in accordance with the SS-data stored against BS group 2x.

When TS61 is requested in a call set-up and the subscription check for TS61 is negative, but a subscription check for TS62 is positive, then the call shall proceed according to the TS 22.003 [6]~~GSM 02.03~~ and TS 27.001 [8~~7~~]. If a subscription check for both TS61 and TS62 is negative, then the call shall be released.

4~~6~~.2 Subscription checking for Supplementary Services

This is described in TS 22.004 [7~~6~~].

Annex A (normative): List of definition of attributes and values used for bearer services

A.1 Information transfer attributes

A.1.1 Information transfer capability

This attribute describes the capability associated with the transfer of different types of information through a ~~GSM~~ PLMN and another network or through a ~~GSM~~ PLMN.
Values:

- unrestricted digital information;
  transfer of information sequence of bits at its specified bit rate without alteration; this implies bit sequence independence, digit sequence integrity and bit integrity.

- speech;
  digital representation of speech information and audible signalling tones of the PSTN coded .

- 3.1 kHz Ex PLMN;
  unrestricted digital information transfer within the PLMN and 3.1 kHz audio restricted within the ISDN.

- Group 3 Fax;
  transfer of Group 3 Fax information.

A.1.2 Information transfer mode

This attribute describes the operational mode of transferring (transportation and switching) through a ~~GSM~~ PLMN.
Values:

- circuit;

~~- packet.~~

A.1.3 Information transfer rate

This attribute describes the bit rate (circuit mode)~~ or the throughput (packet mode)~~. It refers to the transfer of digital information between two access points or reference points.
Values:

- appropriate bit rate, throughput rate.

A.1.4 Structure

This attribute refers to the capability of the ~~GSM~~ PLMN and if involved other networks to deliver information to the destination access point or reference point in a structure NOTE: This attribute has not been utilised in TS 22.002 [5] or TS 22.003 [6].~~GSM 02.03~~.

Values:

- Not applicable.

A.1.5 Establishment of communication

This attribute associated with a telecommunication service describes the mode of establishment used to establish and a given communication.

In every telecommunication service communication may be between users within the ~~GSM~~ PLMN or between a user in the ~~GSM~~ PLMN and a user in another network.

Values:

- demand Mobile Originated (MO) only;
- demand Mobile Terminated (MT) only;
- demand Mobile Originated or Terminated (MO, MT).

A.1.6 Communication configuration

This attribute describes the spatial arrangement for transferring information between two or more access points. It completes the structure associated to a telecommunication services as it associates the relationship between the access points involved and the flow of information between these access points.
Values:

- point-to-point communication;
  this value applies when there are only two access points.

- multipoint communication;
  this value applies when more than two access points (1) are provided by the service. The exact characteristics of the information flows must be specified separately based on functions provided by the ~~GSM~~ PLMN.

  NOTE 1: The number of access points can be undefined.

- broadcast communication;
  this value applies when more than two access points (2) are provided by the service. The information flows are from a unique point (source) to the others (destination) in only one direction.

  NOTE 2: The number of destination access points can be undefined.

A.1.7 Symmetry

This attribute describes the relationship of information flow between two (or more) access points or reference points involved in a communication.
It characterizes the structure associated to a communication service.
Values:

- unidirectional;
  this value applies when the information flow is provided only in one direction.

- bidirectional symmetric;
  this value applies when the information flow characteristics provided by the service are the same between two (or more) access points or reference points in the forward and backward directions.

- bidirectional asymmetric;
  this value applies when the information flow characteristics provided by the service are different in the two directions.

A.1.8 Data compression

This attribute indicates whether use of a data compression function is desired (and accepted) between an MT and IWF.
Values:

- use of data compression requested/not requested;

- use of data compression accepted/not accepted.

A.2 Attributes describing the access at the ~~mobile station~~user equipment

A.2.1 Signalling access

This attribute characterized the protocol on the signalling channel at a given access point or reference point Values:

- manual;
- appropriate V-series protocol;
- appropriate X-series protocol;
- I-series stack of signalling protocols.

A.2.2 Information access

A.2.2.1 Rate

This attribute describes either the bit rate (circuit mode including transparent access to a PSPDN) or variable bit rate (packet mode) used to transfer the user information at a given access point or reference point ~~(access point 1 or 2 at the MS in figure 2/GSM 02.01)~~.

Values:

- appropriate bit rate;

- variable bit rate.

A.2.2.2 Interface

This attribute describes the interface according to the protocol used to transfer user information at a given access point or reference.
Values:

- appropriate V-series DTE/DCE interface;
- appropriate X-series interface;
- S interface;
- analogue 4-Wire interface.

A.3 Interworking attribute

A.3.1 Type of terminating network

Communication can be established between a ~~MS~~UE in a ~~GSM~~ PLMN (originating network) and a terminal in a network (terminating network) including the same ~~GSM~~ PLMN or another ~~GSM~~ PLMN. The attribute designates the terminating network.

NOTE 1: The terms "originating" and "terminating" do not indicate the direction of communication establishment.

NOTE 2: This attribute does not reflect whether there is none, one or several transit networks between the originating and terminating networks.

Values:

- PSTN;
- ISDN;
- PSPDN;
- PDN;
- ~~GSM~~ PLMN;
- Direct access networks.

A.3.2 Terminal to terminating network interface

This attribute describes the interface between a terminal equipment and the terminating network.
Values:

- appropriate V-series (DTE/DCE) interface;

- appropriate X-series interface;
- analogue 2 resp. 4 wire interface;
- S interface (D+B+B).

A.4 General attributes

A.4.1 Supplementary services provided

This attribute refers to the supplementary services to a given telecommunication service.
Values:

- appropriate supplementary services.

A.4.2 Quality of service

The Bearer Services use the Quality of Service attribute to indicate one of the following values:

- transparent;
  service characterized by constant throughput, constant transit delay and variable error rate.
- non-transparent;
  service characterized by an improved error rate with variable transit delay and throughput.

A.4.3 Commercial and operational

A.4.4 Service interworking

Annex B (normative): List of definitions of attributes and values used for teleservices

B.1 High layer attributes

B.1.1 Type of user information

This attribute describes the type of information which the communication offered to the user by the teleservice is based on.
Values:

- speech;
- short message;
- facsimile.

B.1.2 Layer 4 protocol functions

B.1.3 Layer 5 protocol functions

B.1.4 Layer 6 protocol functions

B.1.5 Layer 7 protocol functions

B.2 Low layer attribute (bearer capabilities)

The low layer attributes describe the bearer capabilities which support the teleservice. These low layer attributes and their values are the same as presented in Annex A: List of definitions of attributes and values used for bearer services.

B.3 General attributes

The general attributes are the same as presented in Annex A: List of definitions and values used for bearer services.

Annex C (normative):
Definition of "busy" in a ~~GSM~~ PLMN

C.1 Scope

This annex describes the conditions under which a given mobile subscriber (station) is considered as "busy". In general, this occurs whenever the resources associated with that ~~MS~~UE (and needed to successfully complete the call) exist but are not available for that call. The description is based on the busy definition in the ISDN (ITU-T Recommendation I.221).

In addition, the operation of some Supplementary Services occurs when certain of these resources are busy. Therefore, these "resources busy" are also described herein.

This annex does not cover the cases, when network resources not associated with a given destination are unavailable, or when such resources are out-of-service or otherwise non-functional.

C.2 Network Determined User Busy (NDUB) condition

This condition occurs, when a call is about to be offered, if the information (i.e. traffic) channel ~~(Bm or Lm)~~ is busy and the maximum number of total calls has been reached (see NOTE).

This condition also occurs, when a call is about to be offered and an already on-going call attempt (incoming or outgoing) is in the establishing phase, i.e. not yet active.

When NDUB condition occurs, the PLMN will clear the call and indicate "busy" back towards the calling subscriber (see also section 4).

NOTE: The value of the maximum number of calls is 1 for the basic call. When the supplementary service "Call Waiting" is applicable the value is n+1 where n is the maximum number of calls that can be waiting.

TS 22.135 [11] defines NDUB for Multicall environment.

C.3 User Determined User Busy (UDUB) condition

This condition occurs when a call is offered to a ~~Mobile Station~~user equipment and the ~~MS~~UE responds "user busy" because the subscribers resources (terminal or person using them) are busy. Then the PLMN will clear the call with the indication "busy" back towards the calling subscriber (see also section 4).

C.4 Mobile subscriber busy

A mobile subscriber is considered to be busy if either a "Network Determined User Busy" or a "User Determined User Busy" condition occurs.

Some supplementary services (e.g. Call Forwarding on Busy) may cause the call not to be cleared when a busy condition occurs.

Annex D (normative):
Call setup procedures

D.1 Scope

This annex specifies the service requirements for call setup, both Mobile originated and mobile terminated, in a ~~GSM~~ network, including the establishment of radio contact.

~~For GPRS see TS 22.060 [10].~~

D.2 Mobile Originated Call Setup

When an ~~MS~~UE wishes to start a call and there is no existing radio connection, it requests a signalling channel. When such a signalling channel has been allocated to the ~~MS~~UE, the ~~MS~~UE can transfer the call setup information.

A traffic channel may be allocated at any time before the network informs the ~~MS~~UE that the remote user has answered.

For a call to be set up, certain information needs to be sent by the ~~MS~~UE to the network, defining the call. This information may be provided as default by the ~~MS~~UE, it may be derived from the SIM or be entered by the user either directly into the ~~MS~~UE or from a DTE by using the DTE/DCE Interface.

The following information is sent. Where necessary, default values will generally be inserted by the ~~MS~~UE if not directly specified by the user. The ~~GSM~~ Teleservice Emergency Calls are set up using a special procedure not using the fields described in this section (except for the Bearer Capability.

D.2.1 Called Party Address

This is the address of the called partyusing the TON/NPI specified below. In the case of Dedicated PAD or Packet Access, if NPI is set to PNP, the called party address field may be used to specify the profile to be used. In that case, the address of the called DTE will be given in-band as the second part of two-stage call set-up.

D.2.2 Calling/Called Party Sub-address

This is the sub-address of the calling/called party, in order to provide interworking with ISDN. This is described in more detail in ETS 300 059. Support and use of these fields are optional.

D.2.3 Type of Number

This indicates the format of the called party address. The selection procedure is given in TS 22.030 [9~~8~~]. The following Types of Number are commonly used:

- International Format;
- Open Format ("Unknown");
- Dedicated PAD/Packet Access.

D.2.4 Number Plan Indicator

This indicates the number plan of the called party address. Either of the following number plans may be the "default", depending on the contents of the Called Party Address (TS 22.030 [9~~8~~]):

- ISDN/Telephony E.164;
- Unknown.

Alternatively, one of these number plans may be specified if appropriate:

- Data network X.121;
- Telex network F.69;
- National Numbering Plan;
- Private Numbering Plan.

D.2.5 Bearer Capability

This is used to define the type of call to be set up (telephony, data, rate etc.) For most applications, the ~~MS~~UE will use a set of default conditions, generally on the assumption of a telephony call, unless otherwise set. These may be overridden by the user (or DTE via the DTE/DCE Interface) if desired except for the determination of the channel mode (full or half rate, speech codec conversion).

The ~~MS~~UE shall indicate to the network its channel mode capability in terms of the data channels and the speech codec versions supported.

The network decides which mode to use on the basis of the requested bearer or teleservice, the available network

resources and the channel mode capability of the ~~MS~~UE:
For the “alternate” and “followed-by” services, the same principle applies (with the exception of TS61, where a Full Rate or an Enhanced Full Rate channel shall be provided).
Lower Layer Compatibility and Higher Layer Compatibility Information Elements may also be included.

D.2.6 Calling Line Indication Restriction Override

If the user wishes to override the calling line identification restriction, he may indicate this on a per-call basis as described in TS 22.030 [9~~8~~] and TS 22.081 [10~~9~~].

D.2.7 Action of the Network on Call Setup

On receipt of the call setup message, the network shall attempt to connect the call. However, if insufficient information has been provided by the ~~MS~~UE to indicate the exact Bearer Capability requirements (e.g. due to missing or optional values or for rate adaptation for data), the network may insert the missing information, if this is possible, and the call setup shall proceed using the new information. If the call setup is unsuccessful, the network shall notify the ~~MS~~UE of the cause.

D.3 Mobile Terminated Call Setup

Using the procedures described in ~~GSM~~ TS ~~0~~22.011, the network knows the location area where the ~~MS~~UE is positioned. If the ~~MS~~UE is not already in two way radio communication with the network, the network pages the ~~MS~~UE. Upon receiving its page message, the ~~MS~~UE establishes communication with the selected cell ~~(see GSM 03.22)~~. The network then allocates a channel which is used for signalling and sends call setup information to the ~~MS~~UE.
A traffic channel may be allocated at any instant until just after the call is answered by the ~~MS~~UE.

The network indicates to the ~~MS~~UE that it wishes to offer the ~~MS~~UE a call. This notification includes the proposed bearer capability information, where available (see section D.2.5 above).

D.3.1 Bearer Type

If the calling party specifies the required bearer capability this shall be used for the call setup attempt. If the calling party does not specify the required bearer capability (e.g. because the call originated in the PSTN), the network shall attempt to determine the bearer capability to be used as described below.

The network may use a multi-numbering scheme to define the bearer capability by the MSISDN. In a multi-numbering scheme several MSISDNs are associated with one IMSI. Each MSISDN is used for a different bearer capability. If the network uses a multi-numbering scheme and the calling party has not specified the required bearer capability then the network shall use the bearer capability associated with the called party MSISDN.

The network may use a single-numbering scheme, in which one MSISDN is associated with each IMSI. If the network uses a single-numbering scheme and the calling party has not specified the required service then the network shall omit the bearer capability information.

D.3.2 Response of the ~~MS~~UE

On receipt of the call setup request from the network, the ~~MS~~UE shall check that it is able to support the type of call requested and that it is not User Determined User Busy (see annex C). The ~~MS~~UE then alerts the user.

If the ~~MS~~UE is unable to support the type of call requested, or the information is incomplete, the ~~MS~~UE shall, if possible and not restricted by requirements in other ETSs, reply to the network proposing an alternative set of parameters, indicating those that are different from those proposed by the network. The network then either accepts this new proposal or terminates the call attempt.

D.3.3 Description of Call Re-establishment

Call re-establishment allows the ~~mobile station~~user equipment to attempt to reconnect a call following the loss of radio coverage between the ~~MS~~UE and the network while a call is in progress. Call re-establishment may be initiated by the ~~MS~~UE when it detects this situation, if supported in the network.
Call re-establishment is mandatory in the ME and optional in the network.

Annex E (normative): Automatic calling repeat call attempt restrictions

Call set up attempts referred to in this annex are assumed to be initiated from peripheral equipment or automatically from the MT itself.
A repeat call attempt may be made when a call attempt is unsuccessful for the reasons listed below (as defined in GSM 04.08 [12]).
These reasons are classified in three major categories:

1) "Busy destination":

    Cause number 17 User busy.

2) "Unobtainable destination - temporary":

Cause number 18 No user responding;

19 User alerting, no answer;

27 Destination out of order;

34 No circuit/channel available;

41 Temporary failure;

42 Switching Equipment congestion;

44 Requested circuit/channel not available;

47 Resources unavailable, unspecified.

3) "Unobtainable destination - permanent/long term":

Cause number 1 Unassigned (unallocated) number;

3 No route to destination;

22 Number changed;

28 Invalid number format (uncompleted number);

38	Network out of order.

NOTE 1: Optionally, it is allowed to implement cause number 27 in Category 3, instead of Category 2, as this is desirable already in Phase 1.

The table below describes a repeat call restriction pattern to any B number. This pattern defines a maximum number (n) of call repeat attempts; when this number n is reached, the associated B number shall be blacklisted by the MT until a manual re-set at the MT is performed in respect of that B number. When a repeat attempt to anyone B number fails, or is blacklisted, this does not prevent calls being made to other B numbers.
For the categories 1 and 2 above, n shall be 10; for category 3, n shall be 1.

call attempts	**Minimum duration between Call attempt**
Initial call attempt	-
1st repeat attempt	5 sec
2nd repeat attempt	1 min
3rd repeat attempt	1 min
4th repeat attempt	1 min
5th repeat attempt	3 min
	.
	.
nth repeat attempt	3 min

The number of B numbers that can be held in the blacklist is at the manufacturers discretion but there shall be at least 8. However, when the blacklist is full the MT shall prohibit further automatic call attempts to any one number until the blacklist is manually cleared at the MT in respect of one or more B numbers.

When automatic calling apparatus is connected to an MT1 or MT2, or where an MTO is capable of auto-calling, then the MT shall process the call requests in accordance with the sequence of repeat attempts defined above, i.e. requests for repeat attempts with less than the minimum allowed duration between them shall be rejected by the MT.

A successful call attempt to a number which has been subject to the call restrictions shown above (i.e. an unsuccessful call set up attempt has previously occurred) shall reset the "counter" for that number.

The "counter" for an unsuccessfully attempted B number shall be maintained in 24 hours or until the MT is switched off.

The automatic calling repeat call attempt restrictions apply to speech and data services.

NOTE 2: The restrictions only apply to unsuccessful Call Control activity, not to Radio Resource Management or to Mobility Management, so multiple attempts at radio channel access are not limited by this mechanism.

Annex F(normative): Procedures for call progress indications

F.1 General

Indications of call progress, such as ringing, engaged, unobtainable, and no radio channel, may in principle be verbal message, tones, displayed text or graphical symbols. Which combination of these applies may depend on the message, the UE and selection by the user or PLMN operator. However, verbal announcements will generally be reserved for situations which are peculiar to a mobile network, where users may be unfamiliar with any tone chosen to indicate conditions such as "call diversion" or "subscriber not available".
It may also be desirable to add comfort indications (e.g. tones, noise, music, clicks) while a call is being connected, since silence may cause an unfamiliar user to believe that nothing is happening.
Generally, on data calls, and on the data part of alternate speech/data or speech-followed-by-data calls, PLMN generated network tones and announcements should be muted.

F.2 Supervisory tones

F.2.1 General

Supervisory Tones, indicating primarily ringing, engaged and unobtainable numbers, may be generated by both the PLMN and PSTN.

Except for ring tone, all tones indicating call progress to a user shall be generated in the UE, on the basis of signals from the network where available, and are according to the standard defined in the present document.

Tones sent to a caller to a UE will be generated in the network, generally local to the caller, and will be to the standard of his local exchange, except for mobile to mobile calls, where the tones will be generated in the calling UE. For mobile terminated calls, the ring tone will be generated in the called MSC (except OACSU).

F.2.2 Method

In the interests of early release of the traffic channel on failure to succeed in setting up a (mobile originated) call, where possible supervisory tones should be indicated over signalling channels. The UE will then generate the required tones. However, if the network generates an in-band announcement this will be indicated to the UE. In this case the UE shall connect the user to the announcement until instructed to release the call, either by the user or by the network. An alternate procedure may apply for UE able to generate appropriate announcements internally.

The ring tone will be sent over the traffic channel, since this channel must be available for traffic immediately it is answered (exception: Off Air Call Set Up). The Ring Tone is therefore generated by the PLMN or PSTN supporting the called phone.

On failed mobile terminated call attempts, the called MSC will either signal to the caller, if this is possible, or else will generate the required supervisory tones.

"Alert" is not a supervisory tone. The indication is signalled, and the UE may generate any form of indication to the user that the UE is being called.

F.2.3 Standard tones

UE generated tones will be generally in accordance with CEPT (GSM), or ANSI T1.607 (PCS 1900) recommendations, where appropriate, and are listed in table 1. Any network originated tones will be according to PLMN or PSTN choice.

F.2.4 Applicability

This method will apply in all cases where signalling is capable of indicating the supervisory tone required. However, for connection to certain fixed networks where this signalling is not possible, fixed network tones will be carried over the traffic channel.
User equipment may employ any suitable technique to indicate supervisory information. However, if tones are employed, they shall be in accordance with the present document. The use of these tones in the MSC is preferred.

NOTE 1: The tones and/or announcement to the calling party should not be provided if the Information transfer capability is set to UDI.

NOTE 2: For a call with information transfer capability set to 3.1 kHz, the use of tones and/or announcement may cause the expiry of an awaiting answer timer in a modem or fax machine.

F.2.5 Comfort tones

If desired by the PLMN operator, the network may optionally introduce "comfort tones" while the call is being connected, during what would otherwise be silence. This would be overridden by indication of a supervisory tone, an announcement or by traffic. PLMNs may offer this feature optionally to incoming or outgoing callers.

The "comfort tones" may take the form of tones, clicks, noise, music or any other suitable form, provided that they cannot be confused with other indications that might be expected.
This feature is intended to indicate to the user that his call is progressing, to prevent him terminating the call prematurely.

Table 1: Supervisory tones in UEs

Tone		**Frequency**		**Tolerance**	**Type**	
		CEPT	**ANSI**		**CEPT**	**ANSI**
1	Dial tone (optional)	425Hz	350Hz added to 440Hz	15Hz	Continuous	Continuous
2 *	Subscriber Busy (Called Number)	425Hz	480Hz added to 620Hz	15Hz	Tone on 500ms Silence 500ms	Tone on 500ms Silence 500ms
3 *	Congestion	425Hz	480Hz added to 620Hz	15Hz	Tone on 200ms Silence 200ms	Tone on 250ms Silence 250ms
4	Radio Path Acknowledgement (Mobile Originated only) (optional)	425Hz	425Hz	15Hz	Single tone 200ms	Single tone 200ms
5	{Radio Path Not Available {Call Dropped – Mobile originated only	425Hz	425Hz	15Hz	200ms} On/off 200ms} for 3 burst	200ms} On/off 200ms} for 3 burst
6 *	Error/Special Information} Number Unobtainable } Authentication Failure }	950Hz 1400Hz 1800Hz	950Hz 1400Hz 1800Hz	50Hz 50Hz 50Hz	{Triple Tone {Tones on 330ms {Silence 1.0s	{Triple Tone {Tones on 330ms {Silence 1.0s
7	Call Waiting Tone	425 Hz (tolerance 15Hz), on for 200ms, off for 600ms on for 200ms, off for 3s, on for 200ms, off for 600ms on for 200ms. This tone is superimposed on the audio traffic received by the called user. Alternate tones are *acceptable* but not preferred.				

			440 Hz, on for 300 ms, 9.7s off followed by (440 Hz, on for 100 ms off for 100 ms, on for 100 ms, 9.7s off and repeated as necessary) This tone is superimposed on the audio traffic received by the called user.		
Definition of these and other tones, together with advice on announcements, may be found in CEPT T/CS 20-15 and in T/SF 23.					
* The duration of these tones is an implementation option. However, in each case, the UE should be returned immediately to the idle state, and will be able to originate/receive calls, which will override these tones.					
Ringing Tone (Alternative National options permitted)	425Hz	440Hz added to 480Hz	15Hz	Tone on 1s Silence 4s	Tone on 2s Silence 4s
For application of Call Control Cause Information Elements to these tones, see F.4.					

F.3 Recorded announcements

In present networks, both fixed and cellular, the language of recorded announcements and displayed information is invariably that of the country of origin. However, this is generally undesirable in a multi-lingual environment such as is encountered on a global network with international roaming. It is therefore probably desirable to minimise the number of such announcements.

Advanced UEs may be designed which have the ability to generate announcements in the form desired by the user, e.g. in the language preferred by the user. In this case, it becomes necessary to block any verbal announcements sent from the network towards the UE, to avoid clashes with those generated by the UE. The UE may be allowed to block in-band announcements in case appropriate announcements according to the Cause Information Elements (F.3) can be generated. The default setting of the UE shall be "non blocking", which could be set by MMI command to "blocking".

Announcements generated by the PLMN and sent to callers to that PLMN will generally be in the language of the PLMN. However, on some fixed networks it will be possible for the message to be signalled back to the caller's local exchange, which will then generate the announcement in its local language.

F.4 Application of call control cause information elements to supervisory tones

The Cause Information Elements are listed and defined in GSM 04.08 [13]. This annex lists these elements and indicates which supervisory tone should be generated in response. It should be noted that some conditions (e.g. radio path not available, dropped call) may be deduced by the UE, rather than signalled explicitly over the air interface. All causes not listed below should result in the generation of tone 6. In case of multiple calls a tone should only be generated if it does not disturb an ongoing active call. "-" indicates no tone required.

Cause CC		Tone (see table 1)
16	Normal Clearing	1
17	User Busy	2
22	Number Changed	-
30	Response to STATUS ENQUIRY	-
31	Normal, unspecified	-
34	No circuit/channel available	3
41	Temporary Failure	3
42	Switching Equipment Congestion	3
44	Requested circuit/channel not available	3
49	Quality of Service Unavailable	3

58	Bearer Capability not available	3

3G CHANGE REQUEST

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

22.003 **CR** **001** Current Version: 3.0.0

3G specification number ↑ *↑ CR number as allocated by 3G support team*

For submision to TSG SA#6 for approval X
list TSG meeting no. here ↑ for information
(only one box should be marked with an X)

Form: 3G CR cover sheet, version 1.0 The latest version of this form is available from: ftp://ftp.3gpp.org/Information/3GCRF-xx.rtf

Proposed change affects: USIM X ME X UTRAN X Core Network X
(at least one should be marked with an X)

Source: SMG1/ SA WG1 **Date:**

Subject: Editorial update for GSM/3GPP use.

3G Work item:

Category:	F	Correction	
	A	Corresponds to a correction in a 2G specification	
(only one category	B	Addition of feature	
shall be marked	C	Functional modification of feature	X
with an X)	D	Editorial modification	

Reason for change: The transfer of GSM specifications for 3GPP requires an editorial update. Text referring to the GSM system needs to be changed to refer to both the GSM and 3G systems. 02.03 is proposed to be transferred to 22.003 with this CR. 02.03 describes only the CS domain requirements.

Clauses affected: All clauses

Other specs affected:	Other 3G core specifications		→ List of CRs:
	Other 2G core specifications		→ List of CRs:
	MS test specifications		→ List of CRs:
	BSS test specifications		→ List of CRs:
	O&M specifications		→ List of CRs:

Other comments:

help.doc

<--------- double-click here for help and instructions on how to create a CR.

~~ETSI TS 100 905 V7.0.0+643~~ TS 22.003

Technical Specification (1999-08

Digital cellular telecommunications system (Phase 2+);
Circuit t~~T~~eleservices supported by a
Public Land Mobile Network (PLMN)
(~~GSM 0~~TS 22.003 version 3~~7~~.0.0 Release 1998)

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0 Scope

The present document describes and defines a recommended set of circuit t~~T~~eleservices to be supported by a ~~GSM~~ PLMN in connection with other networks as a basis for defining the network capabilities required.

~~Teleservices not included in the present document should not be introduced unilaterally by a mobile network operator, if the provision of this service requires that the GSM signalling specifications are modified.~~

0.1 Normative references

This ETS incorporates by dated and undated reference, provisions from other publications. These normative references are cited at the appropriate places in the text and the publications are listed hereafter. For dated references, subsequent amendments to or revisions of any of these publications apply to this ETS only when incorporated in it by amendment or revision. For undated references, the latest edition of the publication referred to applies.

[1] GSM 01.04 (ETR 350): "Digital cellular telecommunications system (Phase 2+); Abbreviations and acronyms".

[2] TS~~GSM~~ 2~~0~~2.001: "~~Digital cellular telecommunications system (Phase 2+);~~ Principles of circuit telecommunication services supported by a ~~GSM~~ Public Land Mobile Network (PLMN)".

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

[4] TS 22.004: " General on supplementary services".

[5] GSM ~~TS~~ 022.~~0~~68 : " Digital cellular telecommunications system (Phase 2+); Voice Group Call Service (VGCS) - Stage 1".

[6] GSM ~~TS~~ 2~~0~~2.~~0~~69 : " Digital cellular telecommunications system (Phase 2+); Voice Broadcast Service (VBS) - Stage 1".

[7] TS 23.040: " Technical realization of the Short Message Service (SMS) Point-to-Point (PP)".

[8] TS 23.041 : " Technical realization of Short Message Service Cell Broadcast (SMSCB)".

[9] GSM 04.08: " Mobile radio interface layer 3 specification".

[10] TS 27.001 : " General on Terminal Adaptation Functions (TAF) for Mobile Stations (MS)".

[11] TS 27.005: " Use of Data Terminal Equipment - Data Circuit terminating Equipment (DTE - DCE) interface for Short Message Service (SMS) and Cell Broadcast Service (CBS)".

[12] ITU-T Recommendation T.4: "Standardization of group 3 facsimile apparatus for document transmission".

[13] ITU-T Recommendation T.30: "Procedures for document facsimile transmission in the general switched telephone network".

[14] TR 21.905: "Vocabulary for 3GPP Specifications"

[15] TS 22.101: "UMTS Service Principles".

0.2 Abbreviations

Abbreviations used in the present document are listed in GSM 01.04 [1] and TS 21.905[14].

1 Framework for describing circuit teleservices supported ~~by a GSM PLMN~~

Teleservices supported by a ~~GSM~~ PLMN are described by a number of attributes which are intended to be largely independent.

These attributes are described and defined in specification ~~GSM 0~~TS 22.001 [2].

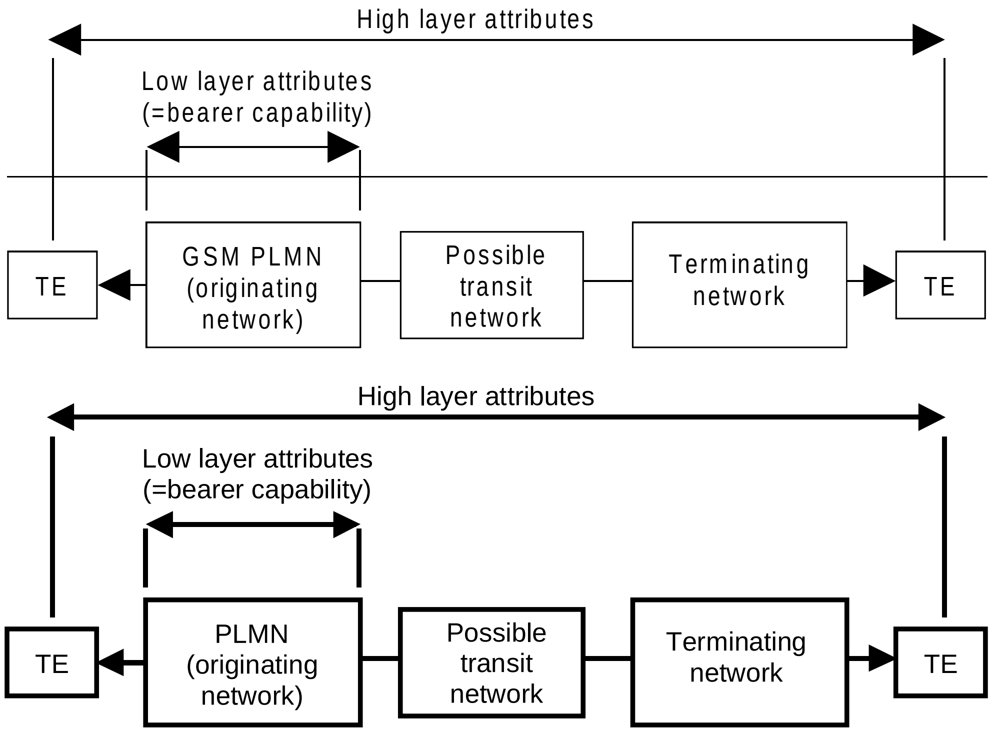
They are grouped into three categories:

- High layer attributes;

- Low layer attributes (describing the Bearer capabilities which support the Teleservice).
    - information transfer attributes;
    - access attributes.
- General attributes.

~~NOTE: Teleservices generally make use of underlying lower layer capabilities of Bearer Services as defined in specification TS 22.002 [3]. However, where Teleservices are provided by a single Administration, RPOA or other services provider, the combination of values of lower layer attributes applicable to specific Teleservices may not necessarily be identical to any of those identified for the Bearer Services appearing in specification TS 22.002 [3].~~

Figure 1 shows the relationship between the different categories of services attributes, and their scope within a Teleservice.



NOTE 1: A transit network may not exist.
NOTE 2: Communication may be established from both ends in principle.

Figure 1: Relationship between the categories of services attributes and their scope within a Teleservice

2 List of the teleservice attributes

Table 1 gives the list of the attributes. For the definitions and possible values of these attributes, see ~~GSM~~ TS 2~~0~~2.001 [2].

Table 1: List of Teleservice attributes

1. High layer capabilities	1.1 Type of user information	Dominant Teleservice attribute category
	1.2 Layer 4 protocol functions	
	1.3 Layer 5 " "	Secondary attributes
	1.4 Layer 6 " "	" "
	1.5 Layer 7 " "	
2. Low layer capabilities (describing the Bearer capability which supports the Teleservice)	2.1 Information transfer	" "
	2.1.1 Information transfer capabilities	" "
	2.1.2 Information transfer mode	" "
	2.1.3 Information transfer rate	Individual services (in the category)
	2.1.4 Structure	
	2.1.5 Establishment of communication	
	2.1.6 Communication configuration	
	2.1.7 Symmetry	
	2.2 Access (~~GSM~~TS ~~0~~22.001)	Qualifying attributes
	2.2.1 Signalling access	" "
	2.2.2 Information access	" "
	2.3 Interworking	" "
	2.3.1 Terminating network type	" "
	2.3.2 National/international interworking	Further specify the individual services
	2.3.3 Interface of terminal equipment	
3. General	3.1 Supplementary services provided	
	3.2 Quality of service	
	3.3 Operational and commercial	

3 List of teleservice categories and individual teleservices

Table 2 presents a list of all Teleservices categories and of individual Teleservices and the associated dominant and secondary attributes.

4 Description of individual teleservices

The annex contains a data sheet per Teleservice with all attributes and comments.

5 Bearer capabilities supporting teleservices

According to specification ~~GSM 0~~TS 22.001 [2] the Bearer Capability defines the technical features of a Teleservice as they appear to the user at the customer access point or an appropriate interface of a fixed network. The Bearer Capability is characterized by information transfer, access and interworking attributes. The same set of attributes as for a Bearer Service is used. A Bearer Capability is associated with every Teleservice.

~~6 Possible further evolution phases of teleservices in a GSM PLMN~~

~~Possible further evolution phases according to the definition in GSM 01.06 could become necessary. For instance, speech coding procedures (half rate speech codec) will provide for the reduction of the bit rate for speech transmission and thus increase the network capacity.~~

Table 2: Teleservice categories and Teleservices

Dominant attribute	**Category of teleservice**		**Individual Teleservice**		
Type of user in- formation	No	Name	No	Name	
Speech	1	Speech trans- mission	11 12	Telephony Emergency Calls	
Short message	2	Short message service	21 22 23	Short message MT/PP Short message MO/PP Short message cell broadcast	
Facsimile	6	Facsimile trans - mission	61	Alternate speech and facsimile group 3	T NT
			62	Automatic Facsimile group 3	T NT
Speech	9	Voice Group service	91 92	Voice Group Call Service Voice Broadcast Service	

~~NOTE: Direct access to private networks is foreseen by recommended provision A.~~

Annex A (normative): Description of individual Teleservices

~~NOTE 1: Interworking with Telex may be provided via teletex-telex- or other interworking functions.~~

NOTE 1~~2~~: Within the ~~GSM~~ PLMN the "Information transfer rate" attribute is not indicated~~, this is because the user may access the PLMN at either an "S" or "R" reference point. In addition, the "Information transfer rate" at other reference points within the PLMN assumed or otherwise may be different.~~

A.1 Individual Teleservices

A.1.1 Telephony

Teleservice 11, Telephony							
	1.	1.1 Type or user information			Speech		
A	HLC	1.2 Layer 4 protocol functions			-		
T		1.3 Layer 5 protocol functions			-		
T		1.4 Layer 6 protocol functions			-		
R		1.5 Layer 7 protocol functions			-		
I	2.	2.1	2.1.1 Information transfer capability		speech (digital representation)		
B	LLC		2.1.2 Information transfer mode		Circuit		
U		Inform	2.1.3 Information transfer rate		not applicable		
T		transfer	2.1.4 Structure		not applicable		
E			2.1.5 Establishment of connection		demand MO MT		
S			2.1.6 Communication configuration		point-to-point		
			2.1.7 Symmetry		bidirectional symmetry		
		2.2	2.2.1 Signalling access		Manual		
		Access	2.2.2 Information access	Rate	full rate/half rate		
		at ~~MS~~UE	(TS~~GSM~~ ~~0~~22.001)	Interface			
		2.3	2.3.1 Visible network type		PSTN/ISDN/ ~~GSM~~ PLMN		
		Inter-	2.3.2 National/Internat. Interworking		international/national		
		working	2.3.3 Interface of TE to terminating		2 wire, analogue	4 wire S (B+B+D)	~~GSM/~~ME
	3.	3.1 Supplementary service provided			TS 22.004		
	Gen	3.2 Quality of service					

Comments:

This service provides the transmission of speech information and audible signalling tones of the PSTN/ISDN. In the ~~GSM~~ PLMN and the fixed network processing technique appropriate for speech such as analogue transmission, echo cancellation and low bit rate voice encoding may be used. Hence, bit integrity is not assured.

1) Transparency for telephone signalling tones is provided.

2) Transparency for voice band facsimile signals is not mandatory. (Appropriate bearer services see TS 22.002 [3].)

3) Transparency for end to end speech encryption is not mandatory. If a user needs to apply this technique an appropriate bearer service (TS 22.002 [3]) can be used.

4) Transmission of DTMF is provided in the mobile to fixed direction (e.g. for controlling voice mail boxes) during any time of an established call.

5) GERAN speech teleservices may be provided using the Full Rate (full rate, version 1), Enhanced Full Rate (full rate, version 2), Half Rate (half rate, version 1) or Adaptive Multirate (AMR) speech codecs. The default speech codec to provide speech service across the GERAN is Full Rate.

6) The default speech codec to provide speech service across the UTRAN is AMR.

A.1.2 Emergency calls

Teleservice 12, Emergency calls						
ATTRIBUTES	1. HLC	1.1 Type or user information			Speech	
		1.2 Layer 4 protocol functions			-	
		1.3 Layer 5 protocol functions			-	
		1.4 Layer 6 protocol functions			-	
		1.5 Layer 7 protocol functions			-	
	2. LLC	2.1 Inform transfer	2.1.1 Information transfer capability		speech (digital representation)	
			2.1.2 Information transfer mode		Circuit	
			2.1.3 Information transfer rate		not applicable	
			2.1.4 Structure		not applicable	
			2.1.5 Establishment of connection		demand MO MT	
			2.1.6 Communication configuration		point-to-point	
			2.1.7 Symmetry		bidirectional symmetry	
		2.2 Access at ~~MS~~UE	2.2.1 Signalling access		Manual	
			2.2.2 Information access (~~GSM 02.01~~TS 22.001)	rate	full rate/half rate	
				interface		
		2.3 Inter-working	2.3.1 Visible network type		PSTN	ISDN
			2.3.2 National/Internat. interworking		national	
			2.3.3 Interface of TE to terminating Ntwk.		2 wire	4 wire
	3. Gen	3.1 Supplementary service provided			TS 22.004 (see note 3)	
		3.2 Quality of service				

Comments:

1) A standardized access method throughout all ~~GSM~~ PLMNs is mandatory. ~~In addition national emergency call numbers of PSTN/ISDN must be usable from MS.~~See TS 22.101[5] for further information on emergency call requirements.

2) It shall be an option of the network operator whether to accept emergency calls coming from ~~mobile station~~user equipment~~s~~ which do not transmit an IMSI or a TMSI.

3) Emergency calls supersede all constraints imposed by supplementary services or ~~mobile station~~user equipment features used for other Tele or Bearer services. The lock state of the ~~MS~~UE is overridden by the SOS-procedure.

4) Emergency calls will be routed to the emergency services in accordance with national regulations.

5) In order to help identifying callers in cases of misuse databases in the ~~GSM~~ PLMN may be accessed to retrieve the identity of the calling ~~MS~~UE.

A.1.3 Short Message Service (SMS)

A.1.3.1 Short message service MT/PP

Teleservice 21, Short Message MT point-to-point 1), 2)					
	1.	1.1 Type or user information			short message, ≤ 160 characters
A		1.2 Layer 4 protocol functions			
T		1.3 Layer 5 protocol functions			see TS 23.040
T		1.4 Layer 6 protocol functions			see TS 23.040
R		1.5 Layer 7 protocol functions			see TS 23.040
I	2.	2.1	2.1.1 Information transfer capability		not applicable
B			2.1.2 Information transfer mode		not applicable
U		Inform	2.1.3 Information transfer rate		not applicable
T		Transfer	2.1.4 Structure		not applicable
E			2.1.5 Establishment of connection		not applicable
S			2.1.6 Communication configuration		not applicable
			2.1.7 Symmetry		not applicable
		2.2	2.2.1 Signalling access		see TS 27.005
		Access	2.2.2 Information access	rate	not applicable
		at ~~MS~~UE	(~~GSM 02.01~~TS 22.001)	interface	
		2.3	2.3.1 Visible network type		not applicable 3)
		Inter-	2.3.2 National/Internat. interworking		not applicable 3)
		Working	2.3.3 Interface of TE to terminating Ntwk.		not applicable 3)
	3.	3.1 Supplementary service provided			TS 22.004
	Gen	3.2 Quality of service			

Comments:

1) This service provides the transmission of a short message from a message handling system (service centre) to a ~~mobile station~~ user equipment. The service centre is functionally separated from the ~~GSM~~ PLMN.

2) After reception an acknowledgement message should be sent back.

3) There is only an interworking between the PLMN and SMS Service Centre (SMS-SC). Connections from the fixed network to the SMS-SC are out of the scope of the ~~GSM Standard~~3GPP specifications.

4) The information transfer attributes refer to the connection-oriented services (ISDN, Bluebook Q.931). The Short Message Service is not a connection orientated service, hence the transfer attributes here are not applicable.

5) SMS MT/PP teleservice can be provided via both the CS and PS domains.

A.1.3.2 Short message service MO/PP

Teleservice 22, Short Message MO point-to-point 1), 2)					
	1.	1.1 Type or user information			short message, ≤ 160 characters
A		1.2 Layer 4 protocol functions			
T		1.3 Layer 5 protocol functions			see TS 23.040
T		1.4 Layer 6 protocol functions			see TS 23.040
R		1.5 Layer 7 protocol functions			see TS 23.040
I	2.	2.1	2.1.1 Information transfer capability		not applicable
B			2.1.2 Information transfer mode		not applicable
U		Inform	2.1.3 Information transfer rate		not applicable
T		transfer	2.1.4 Structure		not applicable
E			2.1.5 Establishment of connection		not applicable
S			2.1.6 Communication configuration		not applicable
			2.1.7 Symmetry		not applicable
		2.2	2.2.1 Signalling access		see TS 27.005
		Access	2.2.2 Information access	rate	not applicable
		at ~~MS~~UE	(~~GSM 02.01~~TS 22.001)	interface	
		2.3	2.3.1 Visible network type		not applicable 3)
		Inter-	2.3.2 National/Internat. interworking		not applicable 3)
		working	2.3.3 Interface of TE to terminating Ntwk.		not applicable 3)
	3.	3.1 Supplementary service provided			TS 22.004
	Gen	3.2 Quality of service			

Comments:

1) This service provides the transmission of a short message from a ~~mobile station~~user equipment to a message handling system (service centre). The service centre is functionally separated from the ~~GSM~~ PLMN.

2) After reception an acknowledgement message is sent back.

3) There is only an interworking between the PLMN and SMS Service Centre (SMS-SC). Connections from the fixed network to the SMS-SC are out of the scope of the ~~GSM Standard~~3GPP specifications.

4) The information transfer attributes refer to the connection-oriented services (ISDN, Bluebook Q.931). The Short Message Service is not a connection orientated service, hence the transfer attributes here are not applicable.

5) Information from the following sources at the ~~MS~~UE might be transmitted:

- a pre-recorded message in a store;
- a number from the dialling key pad;
- information from an external keyboard or terminal equipment connected to the ME.

6) SMS MO/PP teleservice can be provided via both the CS and PS domains.

A.1.3.3 Short message service Cell Broadcast (CB)

Teleservice 23, Short Message transmission cell broadcast					
	1.	1.1 Type or user information			short message, ≤ 93 characters 4)
A		1.2 Layer 4 protocol functions			
T		1.3 Layer 5 protocol functions			see TS 23.041
T		1.4 Layer 6 protocol functions			see TS 23.041
R		1.5 Layer 7 protocol functions			see TS 23.041
I	2.	2.1	2.1.1 Information transfer capability		not applicable
B			2.1.2 Information transfer mode		not applicable
U		Inform	2.1.3 Information transfer rate		not applicable
T		transfer	2.1.4 Structure		not applicable
E			2.1.5 Establishment of connection		not applicable
S			2.1.6 Communication configuration		not applicable
			2.1.7 Symmetry		
		2.2	2.2.1 Signalling access		not applicable
		Access	2.2.2 Information access	rate	not applicable
		at ~~MS~~UE	(~~GSM 02.01~~TS 22.001)	interface	not applicable
		2.3	2.3.1 Visible network type		2)
		Inter-	2.3.2 National/Internat. interworking		2)
		working	2.3.3 Interface of TE to terminating Ntwk.		2)
	3.	3.1 Supplementary service provided			TS 22.004
	Gen	3.2 Quality of service			

Comments:

1) This service provides the transmission of a short message from a message handling system to all ~~mobile station~~user equipments in the area of a Base Station. The service centre is functionally separated from the ~~GSM~~ PLMN. There is no acknowledgement message after reception.

2) An interworking only with the Cell-Broadcast Service Centre is foreseen. Connections from the fixed network to the SC are out of the scope of the ~~GSM Standard~~3GPP specifications.

3) The information transfer attributes refer to the connection-oriented services (ISDN, Bluebook Q.931). The Short Message Service is not a connection orientated service, hence the transfer attributes here are not applicable.

4) TS 23.041 provides up to 15 concatenated "pages" of up to 93 characters each.

A.1.3.4 Short message service description

Description of:

teleservice 21, "Short message MT/PP";

teleservice 22 "Short message MO/PP"; and

teleservice 23 "Cell broadcast short messages".

1 Introduction

The purpose of this annex is to describe the short message teleservice.

Three different types of short messages are defined, namely short message MT/PP (Mobile Terminated/Point-to-point), short message MO/PP (Mobile Originated/Point-to-point) and Cell Broadcast short messages.

2 Definition of the short message service MT/PP and MO/PP

For both mobile originated and mobile terminated services the Service Centre acts as store and forward centre. The Service Centre is functionally separate from the PLMN although this does not preclude an integrated implementation. More than one service centre may be connected to a PLMN. Messages may be input to the service centre from a fixed network customer by means of a suitable telecommunications service either from the fixed network, e.g. speech, telex, facsimile, etc. or from a mobile network customer. The list is not intended to be comprehensive and it is entirely open to the service centre provider what telecommunication services it supports. The service centre shall then reformat the message into that provided by the short message service, for delivery to the ~~mobile station~~user equipment.

For mobile originated SMS messages the SMT formats the message into that used by the SMS service and sends to the service centre (to allow interworking with ERMES also ERMES-format addresses may be sent from the ~~MS~~UE to the SC). In general the user may use alphanumeric addresses for more user convenience. In principle the message may be intended for a subscriber on the fixed network or for another mobile subscriber. For the message to another mobile subscriber the service centre should deliver as described in this section.

The message text is limited to a length of 160 characters.

The originator does not need to know the location of the mobile subscriber to whom he wants to send a message. The message is addressed to the recipient's Directory Number.

As a part of the basic service for both MT and MO, an acknowledgement will be provided on a message by message basis to the SC (MT) or ~~MS~~UE (MO). This acknowledgement indicates that the PLMN has successfully transferred the message to the ~~MS~~UE (MT) or SC (MO).

Optionally, the SC may offer final delivery notification to the originator. In this case, the originator may request to have a notification returned from the SC informing her about the delivery of the Short Message to the recipient. This delivery report indicates whether this particular message has been correctly received at the receiving station or not, to the extent that the SC is able to establish this. It does not indicate whether the message has been read. If the delivery report is negative, i.e. the message has not been successfully delivered to the recipient, it shall include the failure cause.

The delivery report is sent to the originator, if reachable, as soon as the information (positive or negative) is available. In addition, the SC may use the delivery report capabilities for other purposes, such as intermediate status reports etc.

All ~~GSM~~ point-to-point short messages are either to or from the service centre. A message from one ~~mobile station~~user equipment to another must pass through a service centre. This case is effectively an MO and MT message together. The two transactions are separate, though clearly related.

Point-to-point messages may be sent or received when the ~~MS~~UE is engaged on a call (voice or data), or in idle mode. However, messages which overlap the boundary of such a call, or during a handover, may be lost, in which case they will be sent again.

The accounting between the SC and PLMN if applicable is for agreement between those parties.

The originator of a short message may notify the SC of an expiry time after which the message is no longer of value and may be deleted by the SC. During the validity period of the message, the SC shall try to deliver the message. After the expiry date the SC will take no further step to deliver the message, but its status may be kept by the SC to enable the originator to enquire the result. If the originator of the short message does not request any expiry time a standard value, e.g. 24 hours, is used.

The Service Centre may give a short message a priority status. This priority message will be attempted to be delivered irrespective of whether or not the ~~MS~~UE has been identified as temporarily absent. Delivery of non-priority messages will not be attempted if the ~~MS~~UE has been identified as temporarily absent.

If necessary, the originator may request the SC to perform specific operations on a previously submitted short message, such as provision/cancellation of a report or deletion of the short message.

The recipient of a short message will be informed by the message about the date and time it was submitted to the SC.

If the ~~MS~~UE Message Store is full, the Message Store Overflow indicator is activated, and any further messages received will not be accepted. An appropriate specific non-acknowledgement message shall be returned. By help of an optional flow control mechanism further waiting short messages will be transmitted after the ~~MS~~UE has memory available again.

3 Reply path

The reply path facility is an enhancement to the point-to-point SMS. In the mobile originated case the mobile user will request his Service Centre to guarantee to forward a single reply to his message back to him (Reply Path).

In the mobile terminated case the recipient of the Short Message will get an indication by the service centre that a reply via this Service centre will be accepted on a subscriptionless basis. The recipient may then submit a reply to this SC (within a period of time defined by the SC operator), which is then forwarded to the submitter of the original message. No subscription with the Service centre is needed by the replying user. The costs, if any, for the reply path are allocated to the originator.

4 Definition of the cell broadcast short message

The cell broadcast is a Teleservice which enables an Information Provider to submit short messages for broadcasting to a specified area within the PLMN.

The cell broadcast service is characterized by the following aspects:

- ~~(i)~~ No acknowledgement is sent from the ~~MS~~UE.

- ~~(ii)~~ The cell broadcast message is sent ~~on control channels~~ in a limited area, defined by the originator of the message, by agreement with the PLMN.

- ~~(iii)~~ An identifier is associated with each message. This identifier is received by the ~~MS~~UE and used by the short message function of the ~~MS~~UE not to store broadcast messages which are not wanted or which have already been received.

~~(iv) Reception is only possible in idle mode.~~

- ~~(v)~~ Generally, cell broadcast messages will be sent continuously, so that all such messages are sent in turn, and then repeated. The cycle time will need to be short enough for important messages to be received by ~~travellers~~ UEs moving through a group of cells.

- ~~(vi)~~ Cell broadcast messages are MT only. The origination of these messages is outside the scope of ~~GSM~~3GPP specifications.

- ~~(vii)~~ The maximum length of each broadcast message (page) will be 1215 characters. A broadcast message can be segmented into pages each with up to 93 characters.

~~The maximum length of each cell broadcast message will be 93 characters. TS 23.041 describes a concatenation mechanism which allows up to 15 of these 93 character messages treated as segments of a longer message. These segments are then referred to as "pages".~~

- ~~(viii)~~ Cell broadcast DRX mode is defined to improve the battery life for ~~Mobile Station~~user equipments. This feature is optional.

- Reception of CBS messages for a UE is not a requirement if it is connected in the CS domain. It should be possible for a UE to receive messages if it is connected in the PS domain and no data is currently transmitted.

~~(ix) The cell broadcast channel allowing the transfer of broadcast messages to the MS is divided into the basic channel and the extended channel. The transfer and scheduling of the messages on both channels shall be done independently. The support of the extended cell broadcast channel by a MS is optional. The reading of the extended SMSCB broadcast channel by the MS shall have low priority, i.e. if necessary the reading of broadcast messages on the extended channel can be interrupted.~~

A.1.4 Alternate speech/facsimile G3

Teleservice 61, Alternate speech and facsimile group 3							
ATTRIBUTES	1. HLC	1.1 Type or user information			facsimile/speech		
		1.2 Layer 4 protocol functions			Procedures according to ITU-T Recommendation T.30/T4.		
		1.3 Layer 5 protocol functions					
		1.4 Layer 6 protocol functions					
		1.5 Layer 7 protocol functions					
	2. LLC	2.1 Inform transfer	2.1.1 Information transfer capability		alternate speech/group 3 fax		
			2.1.2 Information transfer mode		Circuit		
			2.1.3 Information transfer rate		up to 14400 bits/s		
			2.1.4 Structure		not applicable		
			2.1.5 Establishment of connection		demand (MO MT)		
			2.1.6 Communication configuration		point-to-point		
			2.1.7 Symmetry		Bidirectional symmetry		
		2.2 Access at ~~MS~~UE	2.2.1 Signalling access		I.440/450 (GSM 04.08)		
			2.2.2 Information access (~~GSM 02.01~~TS 22.001)	rate	Fullrate		
				interface	2 wire analogue		
		2.3 Inter-working	2.3.1 Visible network type		PSTN	ISDN	PLMN~~GSM~~
			2.3.2 National/Internat. interworking		International/national		
			2.3.3 Interface of TE to terminating		2 wire, analogue/~~GSM MS~~UE		
	3. Gen	3.1 Supplementary service provided			TS 22.004		
		3.2 Quality of service					

Comments:

1) This Teleservice allows the connection of ITU-T group 3 fax apparatus (send and/or receive) to the ~~mobile station~~user equipment~~s~~ of a ~~GSM~~ PLMN. Facsimile connections may be established to/from group 3 apparatus in the PSTN, ISDN or ~~GSM~~ PLMN.

2) A high quality of service even under bad radio conditions and/or in connection to/from moving vehicles is required.

3) Both speech and fax portions of the call will use a full rate. The fax portion of the call may use multiple full rate channels.

4) Subscription for TS61 includes also subscription for TS62 (refer to TS ~~GSM 02.01~~TS 22.001[2]). For this reason and in order to allow a user to change between ME supporting TS61 or TS62 both a network and a ~~MS~~UE supporting TS61 shall also accept call set-ups for TS62. If a subscriber originates/receives a TS61 call but either the ~~MS~~UE or the network do not support TS61 (but supports TS62), then TS61 shall be negotiated to TS62 in accordance to the rules specified in TS 27.001 [10]. If the negotiation does not succeed, then the call shall be released.

A.1.5 Automatic facsimile G3

Teleservice 62, Alternate facsimile group 3							
	1.	1.1 Type or user information			facsimile		
A	HLC	1.2 Layer 4 protocol functions			Procedures according to ITU-T		
T		1.3 Layer 5 protocol functions			recommendation T.30/T4.		
T		1.4 Layer 6 protocol functions					
R		1.5 Layer 7 protocol functions					
I	2.	2.1	2.1.1 Information transfer capability		Facsimile group 3		
B	LLC		2.1.2 Information transfer mode		circuit		
U		Inform	2.1.3 Information transfer rate		up to 14400 bits/s		
T		transfer	2.1.4 Structure		not applicable		
E			2.1.5 Establishment of connection		demand (MO MT)		
S			2.1.6 Communication configuration		point-to-point		
			2.1.7 Symmetry		bidirectional symmetry		
		2.2	2.2.1 Signalling access		I.440/450 (GSM 04.08)		
		Access	2.2.2 Information access	rate	fullrate		
		at ~~MS~~UE	(~~GSM 02.01~~TS 22.001)	interface	2 wire, analogue		
		2.3	2.3.1 Visible network type		PSTN	ISDN	~~GSM~~ PLMN
		Inter-	2.3.2 National/Internat. interworking		international/national		
		working	2.3.3 Interface of TE to terminating		2 wire, analogue/~~GSM~~ ~~MS~~UE		
	3.	3.1 Supplementary service provided			TS 22.004		
	Gen	3.2 Quality of service					

Comments:

1) This teleservice supports a Facsimile Group 3 Autocalling/Autoanswering mode only.

2) This teleservice allows connection of ITU-T group 3 fax apparatus to and from the ~~mobile station~~user equipment~~s~~ of a ~~GSM~~ PLMN. Facsimile connections may be established to and from group 3 apparatus in the PSTN, ISDN or ~~GSM~~ PLMN.

3) A high quality of service even under bad radio conditions and/or in connection to/from moving vehicles is required.

4) If a Network receives a call set-up for TS61 and if the subscriber in question has a subscription for TS62 only, then the network shall negotiate TS61 to TS62 in accordance to the rules specified in TS 27.001 [10]. If the negotiation does not succeed, then the call shall be released. See also item 4) in the description of TS61.

5) This teleservice may use ~~multiple~~ the multislot mechanism of GERAN~~full rate channels~~.

A.1.6 Voice Group Call Service

Teleservice 91, Voice Group Call Service							
	1.	1.1 Type or user Information			Speech		
A	HLC	1.2 Layer 4 protocol functions			-		
T		1.3 Layer 5 protocol functions			-		
T		1.4 Layer 6 protocol functions			-		
R		1.5 Layer 7 protocol functions			-		
I	2.	2.1	2.1.1 Information transfer capability		speech (digital representation)		
B	LLC		2.1.2 Information transfer mode		Circuit		
U		Inform	2.1.3 Information transfer rate		not applicable		
T		transfer	2.1.4 Structure		not applicable		
E			2.1.5 Establishment of connection		demand MO MT		
S			2.1.6 Communication configuration		Multipoint		
			2.1.7 Symmetry		bidirectional symmetry		
		2.2	2.2.1 Signalling access		Manual		
		Access	2.2.2 Information access	Rate	full rate/half rate		
		at ~~MS~~UE	(~~GSM 02.01~~TS 22.001)	Interface			
		2.3	2.3.1 Visible network type		PSTN/ISDN/~~GSM~~ PLMN		
		Inter-	2.3.2 National/Internat. Interworking		international/national		
		working	2.3.3 Interface of TE to terminating		2 wire, analogue	4 wire S (B+B+D)	~~GSM/~~ME
	3.	3.1 Supplementary service provided			TS 22.068		
	Gen	3.2 Quality of service					

Comments:

This service provides for speech conversation of a predefined group of service subscribers in half duplex mode on the radio link taking into account multiple mobile service subscribers involved in the VGCS call per cell. A detailed service description is given in GSM ~~TS 2~~02.~~0~~68 [5].

This teleservice shall only be provided via a GERAN.

A.1.7 Voice Broadcast Service

Teleservice 92, Voice Broadcast Service							
	1.	1.1 Type or user Information			Speech		
A	HLC	1.2 Layer 4 protocol functions			-		
T		1.3 Layer 5 protocol functions			-		
T		1.4 Layer 6 protocol functions			-		
R		1.5 Layer 7 protocol functions			-		
I	2.	2.1	2.1.1 Information transfer capability		speech (digital representation)		
B	LLC		2.1.2 Information transfer mode		circuit		
U		Inform	2.1.3 Information transfer rate		not applicable		
T		transfer	2.1.4 Structure		not applicable		
E			2.1.5 Establishment of connection		demand MO MT		
S			2.1.6 Communication configuration		broadcast		
			2.1.7 Symmetry		unidirectional		
		2.2	2.2.1 Signalling access		manual		
		Access	2.2.2 Information access	Rate	full rate/half rate		
		at ~~MS~~UE	(~~GSM 02.01~~TS 22.001)	Interface			
		2.3	2.3.1 Visible network type		PSTN/ISDN/~~GSM~~ PLMN		
		Inter-	2.3.2 National/Internat. Interworking		international/national		
		working	2.3.3 Interface of TE to terminating		2 wire, analogue	4 wire S (B+B+D)	~~GSM/~~ME
	3.	3.1 Supplementary service provided			TS 22.069		
	Gen	3.2 Quality of service					

Comments:

—This service provides for the distribution of speech, generated by a service subscriber, to all or a predefined group service subscribers located in this area. A detailed service description is given in ~~TS 2~~GSM 02.~~0~~69 [6].

This teleservice shall only be provided via a GERAN.

TSG-SA Working Group 1 meeting #6 **TSG S1** ***(99) 1029***

San Diego, 29 Nov – 03 Dec 1999 Agenda: 6.0

3G CHANGE REQUEST

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

22.030 CR 005 Current Version: 3.1.0

3G specification number ↑ *↑ CR number as allocated by 3G support team*

For submision to TSG	SA#6	for approval	X	*(only one box should*
list TSG meeting no. here ↑		for information		*be marked with an X)*

Form: 3G CR cover sheet, version 1.0 The latest version of this form is available from: ftp://ftp.3gpp.org/Information/3GCRF-xx.rtf

Proposed change affects: *(at least one should be marked with an X)*	USIM	X	ME	X	UTRAN		Core Network	

Source:	SA WG1	**Date:**	
Subject:	Mainly an editorial update for GSM/3GPP use.		
3G Work item:			

Category:			
	F	Correction	
	A	Corresponds to a correction in a 2G specification	
(only one category	B	Addition of feature	
shall be marked	C	Functional modification of feature	X
with an X)	D	Editorial modification	

Reason for change:	The transfer of GSM specifications for 3GPP requires an editorial update. Text referring to the GSM system needs to be changed to refer to both the GSM and 3G systems. This specification includes MMI requirements relevant to ~~CS Domain~~circuit services.
Clauses affected:	All clauses

Other specs affected:			
	Other 3G core specifications		→ List of CRs:
	Other 2G core specifications		→ List of CRs:
	MS test specifications		→ List of CRs:
	BSS test specifications		→ List of CRs:
	O&M specifications		→ List of CRs:

Other comments:	

help.doc

<--------- double-click here for help and instructions on how to create a CR.

3G TS 22.030 V3.1.0 (1999-1(

Technical Specificat

3rd Generation Partnersl
Technical Specification Group Services and System Aspects;
Man-Machine Interface (MMI) of the ~~Mobile Station~~User Equipment (UE~~MS~~)
(3G TS 22.030 version 3.1.0)

0 Scope

This TS defines the requirements for and gives guidelines on the MMI ~~for calls~~ on the User Equipment~~Mobile Station~~ (UE~~MS~~). This includes the requirements of the user procedures for call control and supplementary service control, the requirements on the physical input media and the output, such as indications and displayed information.
This specification included requirements only to UE connected to CS Domain. See TS 22.101[19]; for overall service principles and TS 22.001[20] for Circuit telecommunication services.
~~This TS complements specifications GSM 02.07 [3], TS 22.011 [4], 02.17 [5], 02.40 [7], 03.01[11], TS 23.009 [12], TS 23.012 [13], TS 23.014 [14], TS 24.008 [16], 05.08 [18], and 11.10 [20] and deals with MMI items not covered by these specifications.~~

~~Note: The present document covers description for GSM only. The document needs to be updated to make it applicable to 3GPP.~~

0.1 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.
- For this Release 1999 document, references to GSM documents are for Release 1999 versions (version 8.x.y).

[1] GSM 01.04: "Digital cellular telecommunication system (Phase 2+); Abbreviations and acronyms".

[2] TS 22.004: " General on supplementary services".

~~[3] GSM 02.07: "Digital cellular telecommunication system (Phase 2+); Mobile Station (MS) features".~~

[3~~4~~] TS 22.011: " Service accessibility".

~~[5] GSM 02.17: "Digital cellular telecommunication system (Phase 2+); Subscriber identity modules Functional characteristics".~~

[4~~6~~] TS 22.016: " International Mobile station Equipment Identities (IMEI)".

~~[7] GSM 02.40: "Digital cellular telecommunication system (Phase 2+); Procedures for call progress indications".~~

[5~~8~~] TS 22.083: " Call Waiting (CW) and Call Hold (HOLD) supplementary services - Stage 1".

[6~~9~~] TS 22.084: " MultiParty (MPTY) supplementary services - Stage 1".

[7~~10~~] TS 22.090: " Stage 1 description of Unstructured Supplementary Service Data (USSD)".

~~[11] GSM 03.01: "Digital cellular telecommunication system (Phase 2+); Network functions".~~

~~[12] TS 23.009: " Handover procedures".~~

~~[13] TS 23.012: " Location registration procedures".~~

~~[14] TS 23.014: " Support of Dual Tone Multi-Frequency (DTMF) signalling".~~

[8~~15~~] TS 23.038: Alphabets and language".

[9~~16~~] TS 24.008: " Mobile radio interface layer 3 specification; Core Network Protocols – Stage 3".

[10~~7~~] TS 24.080: " Mobile radio interface layer 3 supplementary services specification Formats and coding".

~~[18] GSM 05.08: "Digital cellular telecommunication system (Phase 2+); Radio subsystem link control".~~

[11~~9~~] TS 29.002: " Mobile Application Part (MAP) ".

~~[20] GSM 11.10: "Digital cellular telecommunication system (Phase 2+); Mobile Station (MS) conformity specification".~~

~~[21] GSM 11.11: "Digital cellular telecommunication system (Phase 2+); Specification of the Subscriber Identity Module - Mobile Equipment (SIM - ME) interface".~~

[12~~22~~] GSM 02.81: "Line Identification Supplementary Services – Stage 1"

[13~~23~~] ITU-T Recommendation E.164: "Numbering plan for the ISDN era"~~.~~

~~[24] ITU-T Recommendation E.131: "Subscriber control procedures for supplementary telephone services".~~

[14~~25~~] ITU-T Recommendation E.121: "Pictograms and symbols to assist users of the telephone service"~~.~~

[15~~26~~] TS 22.072: "~~“~~Call Deflection; Stage 1"~~”~~

[16~~27~~] TS 22.091: "~~“~~Explicit Call Transfer Supplementary Service; Stage 1"~~”~~

[17~~28~~] TS 22.093: "~~“~~Call Completion to Busy Subscriber (CCBS); Stage 1"~~”~~

[18~~29~~] TR 21.905: "Vocabulary for 3GPP Specifications"

[19] TS 22.101; "Service principles"

[20] TS 22.001: "Principles of circuit telecommunication services supported by a Public Land Mobile Network (PLMN)"

0.2 Abbreviations

Abbreviations used in this TS are listed in GSM 01.04 [1] and in TR 21.905[18].

0.3 Definitions

Directory Number: A string consisting of one or more of the characters from the set {0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #, a, b, c} associated with a nature of address indicator and number plan indicator. When using the public MMI for the control of supplementary services however, * and # cannot be part of any SC or SI field.

NOTE 1: No such restriction on the SC and SI fields exists when using other (e.g. menu-driven) MMI for the control of supplementary services.

NOTE 2: When using the public MMI, certain limitations on the use of one and two digit directory numbers may apply. The use of other MMI can remove these restrictions.

NOTE 3: This definition is not intended to require the support of all these characters in the MMI itself.

1 General

1.1 Basic philosophy

The basic idea behind this TS is that it should give a minimum level of requirements, with emphasis on items which are seen as important from a ~~pan-European~~ usage point of view. This means, that the requirements are mainly dealing with standardized control procedures of access to services i.e. call establishment, invocation of supplementary services and so on. ~~This also includes standardized network information to the users such as tones and announcements.~~
The requirements on the physical layout of input and output features are kept to a minimum to allow for differentiated types of ~~MS~~UEs and to ease the introduction of future developments in the area of MMI. The standardized control procedures describe the sequence of real actions to be taken by the users. However, since the requirements on the physical input features are minimal the control procedures may differ between ~~MS~~UEs depending on the solution of the manufacturers. The "bridge" between these requirements is however that the same logical actions have to be taken by the user. That is, the user has to provide the same information for the call control and signalling no matter what the method is. This is also valid if an automatic device is used for carrying out the same actions. The logical procedures are therefore defined and standardized in this specification.
~~The drawback of this approach is that the users of Mobile Stations may face a lot of different types of physical MMI which they have to learn. However, to deal with this problem the specification gives a definition of a basic public MMI. The basic public MMI allows non-experienced, casual users to make and receive a telephony call. This definition is directed to manufacturers of public mobile telephones.~~
Optionally, the user may set the ME to disable some or all of the MMI functions defined in this specification. This setting shall only apply when the same SIM is in use ~~(see GSM 02.17 [5] for security policy),~~ otherwise the ME shall enable the standard MMI.

~~1.2 Structure of the specification~~

~~The TS is divided into sections describing respectively the control procedures, the input features and the output features. The final section defines the basic public MMI. For a lot of items for which no particular MMI specification is necessary there is a reference to the specifications where the basic requirements are to be found (i.e. MS features specified in GSM 02.07 [3] and language of announcements specified in GSM 02.40 [7]).~~
2 Physical user input features

2.1 General

This clause gives the requirements or guidelines for the MMI of the input related ~~MS~~UE features. Basic requirements on these features are given in ~~GSM 02.07~~TS 22.101 [19]~~3] and GSM 02.40 [7]~~.

~~2.2 MMI related to MS access~~

~~No requirements additional to those in other GSM specifications (Reference GSM 02.17 [5], 11.10 [20]).~~

2.2~~3~~ MMI related to ~~MS~~UE features

~~The three first issues are covered in GSM 02.07 [3]:~~

* Country / PLMN selection:

The method is manufacturer optional.

* International Access Function ("+" key):

and

* Keypad:

The physical means of entering the characters 0-9, +, * and # (i.e. the SELECT function) may be keypad, voice input device, DTE or other, but there must be means to enter this information.

The relationship on the keypad between the numbers and letters (where used) is important when mnemonic dialling may be used. The following relationship is therefore preferred though optional.

1		6	MNO
2	ABC	7	PQRS
3	DEF	8	TUV
4	GHI	9	WXYZ
5	JKL	0	

~~See also subclause 5.2.1.~~

* ACCEPT, SEND and END functions:

The physical means to perform these functions may be keypad, voice input device, DTE or other, but there must be means to perform these functions. ACCEPT and SEND may use the same means.

* Setting of called Number Fields (Type of Number), use of the "+" key function:

Users may enter a called number in two formats, called here International or Open. The Type of Number (TON) may be set to other values if required, but the procedure for this is not defined here.

"International format":

This is entered by starting with a "+" followed by country code, even for national calls. This method is preferred for roaming and international calls, and highly desirable for storage of short codes or for call-forwarding.

This sets the TON to "International" - see TS 24.008 ~~[16]~~[9].

"Open format":

This is when the "+" is not entered, and the number is entered in the normal way for that network. The number may require a prefix or escape code as normal, for example for entering the international access code or national access code (often "0").

This sets the TON to "Unknown" - see TS 24.008 ~~[16]~~[9]. (This is not the "National" case, which does not permit prefix or escape digits).

Care should be taken with this format, since the dialled number will only be correct in a given network, and may be wrong when roaming. Caution must be applied when using stored numbers or call-forwarding.

* Setting of Called Number Fields (Number Plan Indicator):

The default Number Plan Identification (NPI) shall be ITU-T E.164 ~~[23]~~[13] if all the digits are in the range 0-9 and the NPI shall be "unknown" if other number information is included. However, if the user selects (or has selected) a particular NPI (procedure not defined) then that NPI shall be used.

* Entry of Bearer Capability Information Elements (BCIE):

This is required in order to indicate information such as whether it is a voice or data call, facsimile, synchronous or asynchronous etc. The method for entering this information is of mobile manufacturer's option. For those ~~Mobile Station~~User equipment~~s~~ offering only telephony (and emergency calls), the default BCIE shall be for telephony (or emergency call). For ~~Mobile Station~~User equipment~~s~~ supporting non-voice services, there shall be means to set the BCIE required, by reading the appropriate field in the SIM and possibly otherwise. This field may be associated with or independent of the called number.

~~2.4 MMI related to user information~~

~~These issues are covered in GSM 02.40 [7]:~~

~~* Selection of language of announcements:~~

~~- No additional requirements are defined in this specification.~~

~~2.5 Other input features~~

~~No requirements additional to those in other GSM specifications (Reference TS 23.014 [14] 11.10 [20]).~~

~~3 Indications and output features~~

~~3.1 General~~

~~This clause gives the requirements and guidelines of the MMI aspects of the outputs such as displayed information, indications and tones. Basic requirements on these features are given in GSM 02.07 [3] and GSM 02.40 [7].~~

~~3.2 MMI related to MS access~~

~~No requirements additional to those in other GSM specifications (ref. TS 23.012 [13], TS 24.008 [16], GSM 05.08 [18]).~~

~~3.3 MMI related to MS features~~

~~Country/PLMN Indication:~~

- ~~The country/PLMN Indication (see GSM 02.07 [3] for definition) should be displayed as such that the user can uniquely identify the country and PLMN e.g. by alpha or numeric means or other.~~

~~These issues are covered in GSM 02.07 [3] . No additional requirements are defined in this specification:~~

- ~~Indication of Call Progress Signals.~~
- ~~Display of Called Number.~~
- ~~Short Message.~~
- ~~Call Charge Units Meter.~~

~~3.4 MMI related to user information~~

~~These issues are covered in GSM 02.40 [7]. No additional requirements are defined in this specification:~~

- ~~Selection of language of announcements;~~
- ~~Supervisory tones.~~

~~3.5 Other output features~~

~~No requirements additional to those in other GSM specifications (ref. GSM 03.01 [11], TS 23.009 [12], TS 23.012 [13], TS 24.008 [16] and 05.08 [18]).~~

~~4~~3 Procedures

3~~4~~.1 General

This clause defines the MMI of the service access procedures, and supplementary service control procedures. These procedures are defined as logical procedures and in general no mandatory methods are specified. In order to make the descriptions continuous and clear requirements in TS 22.101 [19]~~GSM 02.07 [3], TS 22.011 [4], 02.17 [5] and 02.40 [7]~~ have been included or are referenced. The mapping between the MMI procedures and the call control entity is specified in TS 24.008 ~~[16]~~[9].

3~~4~~.2 ~~MS~~UE access

The ~~MS~~UE access procedure is comprised of the initial actions the user has to take before calls can be established or received. This procedure includes e.g. insertion of subscriber-card and entering the PIN-code.
As there exist different types of ~~MS~~UE and as requirements in other GSM specifications allow different options the ~~MS~~UE access procedure may differ between ~~Mobile Station~~User equipment~~s~~. The method for describing the ~~MS~~UE access procedures is by using a Mealy-graph, see annex A.
The graph shows the ~~MS~~UE access for simple ~~MS~~UE e.g. hand-held and they may be different for more complex stations. It should also be noted that the exact sequences of events are not described, these may be chosen by the manufacturers. ~~Nevertheless, the related requirements in the other GSM specifications referenced in subclause 4.1 are applicable.~~

3~~4~~.3 Definition of functions

The following functions are applicable and mandatory for the logical procedures for Mobile originated and terminated calls and for the control of Supplementary Services:

ACCEPT: Acceptance of a mobile terminated call.

SELECT: Entry of information.

SEND: Transmission of the entered information to the network.

INDICATION: Call progress indications~~Requirements in GSM 02.40 [7] are applicable~~. Other indications may be given in addition throughout the procedure.

END: Termination of or disconnection from the call. The execution of the END-function may be caused by either party involved in the call by e.g. termination, loss of coverage, invalidation of payment.

34.4 Call Control

34.4.1 General

Voice calls to and from a ~~Mobile Station~~User equipment shall be controlled in accordance with the procedures described below. "Data calls" are expected to be controlled in a similar way but are not here specified.

34.4.2 Voice calls

The voice call is either a normal telephony call or an emergency call.

34.4.2.1 Mobile originated calls

The following sequence of functions shall be used:

SELECT: Entry of called address information.

SEND: Transmission of the called address.

INDICATION: Call progress indications.~~See subclause 4.3.~~

END: Termination of the call.

34.4.2.2 Emergency calls

With ~~Mobile Station~~User ~~s~~equipment supporting Telephony, it shall be possible to place an emergency call as specified in TS 22.101 [19].~~by entering 112 with GSM 900 and GSM 1800, 911 for GSM 1900 in the U.S.A. and Canada, or 08 for GSM 1900 in Mexico, followed by SEND in the manner specified in subclause 4.4.2.1. When a dual or multi-band terminal supporting GSM 1900 and another band is registered on a GSM 900, GSM 1800 or GSM 1900 network, it shall support the initiation of an emergency call by entry of 112, 911 or 08 unless a data call is requested. It may also be possible for a user to enter a preferred emergency MMI code of up to six digits (such as 999) followed by SEND to invoke an emergency call. In this latter case, the preferred code shall be stored in the SIM and the ME recognizes any dialled instance of this code to set up the emergency call. Additional means to place such a call are also allowed, e.g. provision of a dedicated button.~~

~~The MS must support the initiation of an emergency call to "112", "911" for GSM 1900 in the U.S.A. and Canada, or "08" for GSM 1900 in Mexico, without a SIM present in the MS, regardless of the call being accepted or not by the network (national option to require IMSI).~~

~~NOTE: In addition to the above procedure, calls to national emergency services may be made in the way standard for the country of the serving PLMN. However, with the exception of code "112", "911" for GSM 1900 for U.S.A. and Canada, or "08" for GSM 1900 for Mexico, these are not treated within the PLMN as "Teleservice Emergency call" unless the ME recognizes the code as an emergency code as described above, and would require a valid IMSI.~~

34.4.2.3 Mobile terminated calls

The following sequence of functions shall be used:

INDICATION: Alert to the user that she is being called.

ACCEPT: Acceptance of the incoming call by the user.

INDICATION: Call progress indications.~~See subclause 4.3.~~

END: Termination of the call.

User Determined User Busy (UDUB): If, on being alerted by an incoming call, the called user enters "0 SEND", this shall set UDUB for that call, which shall either invoke call forwarding on busy, if active and operative, or else present BUSY to the calling party.

34.5 Supplementary Services Control

34.5.1 General

The supplementary services shall be controlled in accordance with the procedures described below. All ~~Mobile Stations~~User equipments with MMI shall be able to be controlled in this way, to minimize the confusion of users using different types of ~~Mobile Station~~User equipment (quite likely, due to the use of the SIM IC card) and to permit the introduction by a PLMN operator of new supplementary services, not defined at the time of the design of a ~~Mobile Station~~User equipment. These procedures are based on those recommended by ~~CEPT/SF~~ETSI/HF and ITU-T Recommendation E.131.
The specified MMI shall be supported by the L3 signalling between the ~~MS~~UE and the MSC, see TS 24.080 ~~[17]~~[10].
In addition to these specified MMI procedures the ~~MS~~ UE may be equipped with additional enhanced MMI procedures (e.g. dedicated keys, menu procedures...), left to the discretion of the manufacturer. These procedures shall also be converted in accordance with TS 24.080 ~~[17]~~[10].

34.5.2 Structure of the MMI

The following sequence of functions shall be used for the control of Supplementary Services:

SELECT: Entry of the procedure information (may be a digit or a sequence of characters).

SEND: Transmission of the information to the network.

INDICATION: Call progress indications.~~See subclause 4.3.~~

The ~~MS~~UE shall support the MMI procedure specified as:

Activation	:	*SC*SI#
Deactivation	:	#SC*SI#
Interrogation	:	*#SC*SI#
Registration	:	*SC*SI# and **SC*SI#
Erasure	:	##SC*SI#

This structure consists of the following parts:

- Service Code, SC((2 or 3 digits);
- Supplementary Information, SI (variable length).

The procedure always starts with *, #, **, ## or *# and is finished by #. Each part within the procedure is separated by *.
The service code uniquely specifies the Supplementary Service, either as a defined Supplementary Service or as a spare service code. All spare service codes shall be reserved for future use.
The ~~MS~~UE shall determine from the context whether, an entry of a single *, activation or registration was intended.
For example, a call forwarding request with a single * would be interpreted as registration if containing a forwarded-to number, or an activation if not.
The supplementary information (SI) may comprise e.g. a PIN code or Directory Number. Where a particular service request does not require any SI, "*SI" is not entered, e.g. Activation becomes *SC#SEND. Where further supplementary information is required this is again entered as *SI, e.g. *SC*SIA*SIB#SEND. SIB may be used to specify the tele or bearer service expressed as a Basic Service Group to which this supplementary service request applies, SIC may be used to specify the value of the "No Reply Condition Timer".
Use of SIA, SIB, SIC for a particular procedure is optional. The procedure to be adopted where these are not all used is

as follows:
*SI# shall be entered in any of the following formats:

* SIA * SIB * SIC #

* SIA * SIB #

* SIA * * SIC #

* SIA #

* * SIb * SIC #

* * SIB #

* * * SIC #

#

The denotation of the Supplementary Information and the order of entry are specified in annex B. Supplementary Information Codes for the Teleservices and Bearer Services are given in annex C.
The following procedures shall be used for application of supplementary services to the call set-up procedure:

*SCn*SI#DN SEND;

where SC is the service code defined in annex B and *SI is an optional field which may be applicable to service SC. The "n" is a single digit used to indicate the numbering plan, profile, priority, etc. according to the service being applied. For simplicity of presentation, the leading * is shown on the assumption that the action is to activate (switch on) the required service. However, for a deactivation (or switch off), this would become:

#SCn*SI#DN SEND;

It is assumed that the *# (interrogation) will not apply to call set-up.
Where more than one supplementary service is applicable to the call set-up, these shall be concatenated with any applicable supplementary information immediately following the applicable service code.
For example, if SCn and SI refer to one applicable supplementary service and scn and si to another, then the generic procedure becomes:

*SCn*SI#scn*si#DN SEND.

NOTE: The order of entry of SC and sc is a user option, provided that any supplementary information follows immediately after the relevant SC.

Where SI is not applicable according to the definition of the supplementary service, then *SI is omitted. Where its use is optional, but not selected for a particular call set-up, it may be omitted or entered as an extra * if this is necessary to avoid ambiguity of interpretation.

NOTE: By using the # as a separator, most cases are expected to be unambiguous.

34.5.3 Handling of supplementary services

34.5.3.1 Handling of defined supplementary services

The MMI procedure for the defined Supplementary Services (see TS 22.004 [2]) shall be converted to the mobile radio interface Layer 3, as specified in TS 24.080 ~~[17]~~[10]. An appropriate message should be given/displayed to the user in accordance with the "return result/error" from the network.
The service codes for the defined Supplementary Services are given in annex B.

34.5.3.2 Handling of not-implemented supplementary services

The ~~MS~~UE shall act in accordance with figure 34.5.3.2 when digits are entered to the ~~MS~~UE to determine whether to interpret these as call set-up requests or supplementary service control procedures etc.. This may involve a mechanism, referred to as Unstructured SS Data, which allows the support of SS services which are not implemented by means of the specified functional signalling. See also TS 22.090 ~~[10]~~[7].

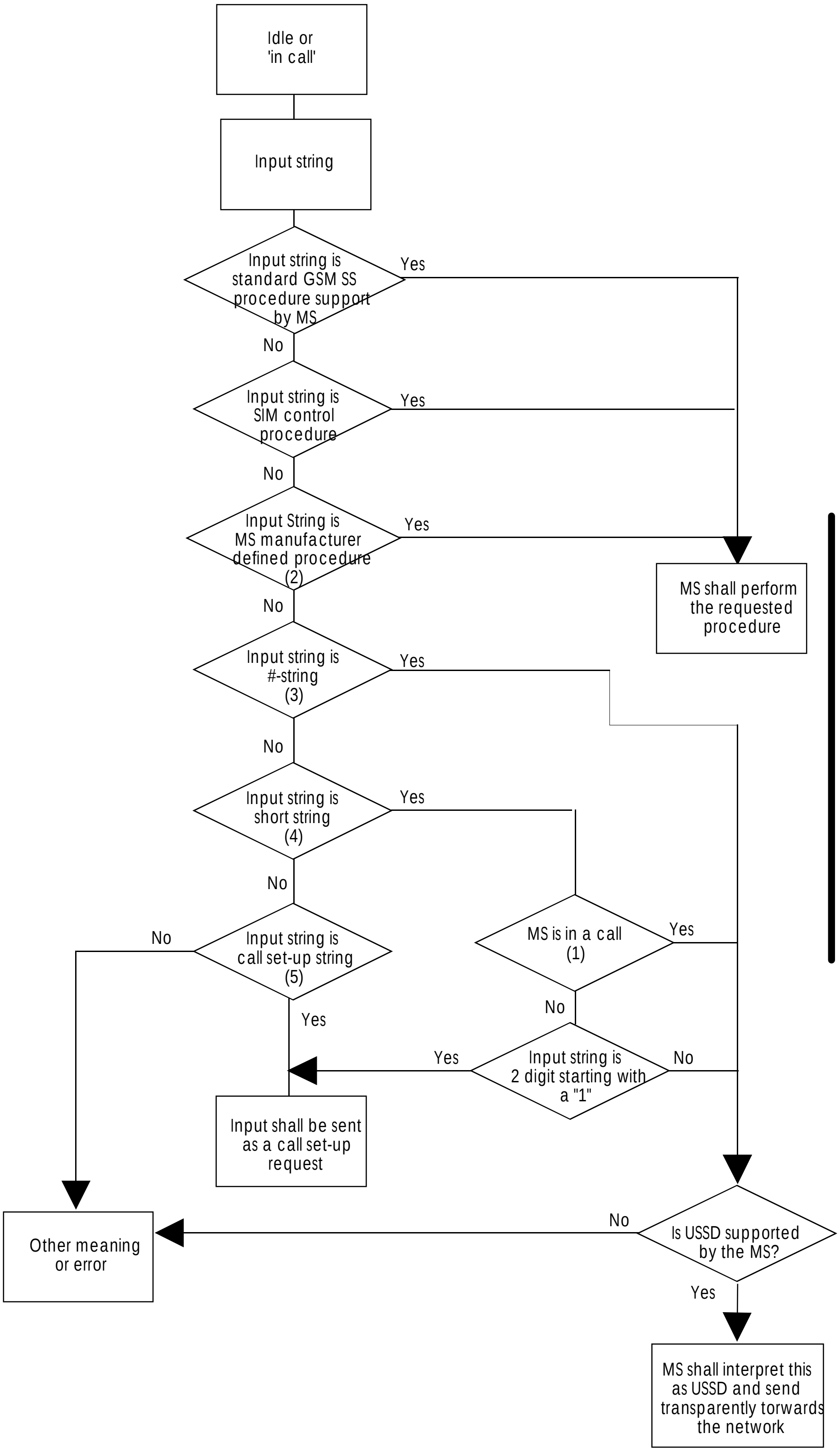


Figure 34.5.3.2

The following definitions are applicable to the interpretation of figure 3~~4~~.5.3.2:

1) In a call:

A ~~MS~~UE is "in a call" from the time that signalling related to the establishment or attempted establishment of a MO or MT call commences and before the call or call attempt ends, and (if applicable) the ME has stopped generating tones related to this call to the user.

2) ~~MS~~UE manufacturer defined procedure:

The term "~~MS~~UE manufacturer defined procedure" shall not include the following two cases:

(i) input which can be interpreted as being of the following form, whether or not in a call:

CX [string]# followed by SEND;

where
string is any combination of numeric digits, *, #;

and
C comprises 1, 2 or 3 digits from the set (*,#);

and
X comprises 1, 2 or 3 numeric digits or the fourth numeric digit is non-zero;

(ii) input of the following form in a call (as defined above):

"Entry of 1 or 2 characters defined in the TS 23.038 ~~[15]~~[8] Default Alphabet followed by SEND".

3) #-string:

Input of the form.

"Entry of any characters defined in the TS 23.038 ~~[15]~~[8] Default Alphabet (up to the maximum defined in TS 24.080 ~~[17]~~[10]), followed by #SEND".

4) Short string:

"Entry of 1 or 2 characters defined in the TS 23.038 ~~[15]~~[8] Default Alphabet followed by SEND".

5) Call setup string:

MMI input in accordance with the call set-up procedures as defined in TS 24.008 ~~[16]~~[9] and terminated by SEND.

If the network has initiated an operation which explicitly (in the signalling) requires a response from the user, then the user shall be able to enter a response in the form of any string of characters followed by SEND. The mobile shall also provide an MMI command to terminate the dialogue with a NULL response.
The use of END shall release all calls in progress (see also subclause 3~~4~~.5.5.2), terminate any outstanding unstructured SS operations, and release any connection used for unstructured SS operations.

3~~4~~.5.4 Registration of new password

The following procedure permits the user to change the password relating to use of Supplementary Services. The only control procedure supported is Registration of a new password, which replaces any previous password for the same service. The password may not be Erased or Interrogated.
Procedure:

* 03 * ZZ * OLD_PASSWORD * NEW_PASSWORD * NEW_PASSWORD #

The ~~MS~~UE shall also support the alternative procedure:

** 03 * ZZ * OLD_PASSWORD * NEW_PASSWORD * NEW_PASSWORD #

where, for Barring Services, ZZ = 330;
for a common password for all appropriate services, delete the ZZ, entering:

* 03 ** OLD_PASSWORD * NEW_PASSWORD * NEW_PASSWORD #

The ~~MS~~UE shall also support the alternative procedure:

** 03 ** OLD_PASSWORD * NEW_PASSWORD * NEW_PASSWORD #

the ~~MS~~UE will then indicate to the user whether the new password request has been successful or not. If the new password request is rejected (e.g. due to entry of incorrect old password) the old password remains unchanged, until it is successfully changed by correctly repeating the procedure. Refer to TS 22.004 [2] regarding repeated entry of incorrect password.

NOTE: The procedures shall be followed by SEND as described in subclause 34.5.2.

34.5.5 Handling of supplementary services within a call

34.5.5.1 Call Deflection, Call Waiting, Call Hold, MultiParty Services, Explicit Call Transfer and Completion of Calls to Busy Subscriber general principles

During a call, the following general procedures shall be available, where applicable, for the subscriber to control the operation of

- Call Deflection
- Call Waiting
- Call Hold
- MultiParty Services
- Explicit Call Transfer
- Completion of Calls to Busy Subscriber

including their interactions. It should be noted that not all control procedures described in TS 22.072 ~~[26]~~[15], TS 22.083 ~~[8]~~[5], TS 22.084 ~~[9]~~[6], 22.091 ~~[27]~~[16], and 22.093 ~~[28]~~[17] are specified in this subclause.

Procedures:

Entering 0 followed by SEND	-	Releases all held calls or sets User Determined User Busy (UDUB) for a waiting call.
Entering 1 followed by SEND	-	Releases all active calls (if any exist) and accepts the other (held or waiting) call.
Entering 1X followed by SEND	-	Releases a specific active call X.
Entering 2 followed by SEND	-	Places all active calls (if any exist) on hold and accepts the other (held or waiting) call.
Entering 2X followed by SEND	-	Places all active calls on hold except call X with which communication shall be supported.
Entering 3 followed by SEND	-	Adds a held call to the conversation.
Entering 4 followed by SEND	-	Connects the two calls and disconnects the subscriber from both calls (ECT).
Entering 4 * "Directory Number"	-	Redirect an incoming or a waiting call to the specified followed by SEND directory number.
Entering 5 followed by SEND	-	Activates the Completion of Calls to Busy Subscriber Request.
Entering "Directory Number"	-	Places all active calls (if any exist) on hold and sets up a followed by SEND new call to the specified Directory Number.
Entering END	-	Releases the subscriber from all calls (except a possible waiting call).

"X" is the numbering (starting with 1) of the call given by the sequence of setting up or receiving the calls (active, held

or waiting) as seen by the served subscriber. Calls hold their number until they are released. New calls take the lowest available number.
Where both a held and a waiting call exist, the above procedures shall apply to the waiting call (i.e. not to the held call) in conflicting situation.

34.5.5.2 Call Waiting (CW)

During a call, provided this service is active for the called party, if a second call attempts to make contact, a "call waiting" indication will be presented to the called party.
To clear the current call and accept the waiting call, enter 1 followed by SEND, within the time out period.
Alternatively, either party in the existing, active, call may release that call. The call waiting indication then becomes an "alert", and the call may be accepted as a normal call within the time-out period.
To hold the current call and accept the waiting call, enter 2 followed by SEND, within the time out period.
To ignore the waiting call, take no action.
To set User Determined User Busy (UDUB) for the waiting call, enter 0 followed by SEND, within the time out period.
To redirect the waiting call to another destination, enter 4 * "Directory Number" followed by SEND, within the time out period.

34.5.5.3 Call hold

During a call, the initial call may be held while another call is made by entering the second directory number followed by SEND.
To shuttle between the two calls enter 2 followed by SEND irrespective of whether the second call was acquired using the Call Hold or acceptance of Call Waiting procedures.
If no waiting call exists, by entering 0 followed by SEND the held call is cleared.
To clear an active call and return to the held call enter 1 followed by SEND. This is only possible if no waiting call exists.

34.5.5.4 MultiParty

Having established calls to these two parties with one call active and the other on hold, enter 3 followed by SEND for a multiparty conversation.
To add another remote party, the same procedure applies. Another call is established and either this call or the existing multiparty call is placed on hold. Entering 3 followed by SEND brings all these parties together in an enlarged multiparty call.
To choose one party for a private communication, putting the rest of the multiparty on hold, enter 2X followed by SEND, where X defines the call with which communication shall be supported.
To return to the multiparty, with the previously active call placed on hold, enter 2 followed by SEND.
To release a specific party enter 1X followed by SEND, where X is defined as above.
If the served mobile subscriber enters END, all calls including the multiparty are released. The multiparty is terminated.

34.5.5.5 Explicit Call Transfer

Having established calls to these two parties with one call active and the other on hold, enter 4 followed by SEND to transfer the calls.
If a subscriber has one active, one held and one waiting call, and by entering 4 SEND the active and held call are connected, after the successful completion of the transfer, the served subscriber shall be offered the normal notification that there is a new waiting call, as for a normal terminating call.

34.5.5.6 Special case

Provided both Call Hold and Call Waiting is active, it is possible to have one active and one held call and then a third call attempting to make contact. In this case, to clear the active call and accepting the waiting call (the held call not affected) enter 1 followed by SEND (If entering 2 followed by SEND the call state shall not be affected).
Alternatively, either party in the active call may release that call. The held call will remain held. Within the time-out period the waiting call may then be accepted by entering 2 followed by SEND. It shall also be allowed to accept the waiting call by entering 1 followed by SEND.
As and additional alternative, the (controlling) subscriber B may enter END, in which case the active and the held calls are released. The call waiting indication then becomes an "alert" and the previously waiting call may be accepted as a normal call within the time-out period.

34.5.5.7 Call Deflection

If informed about an incoming call this call may be redirected to an another destination by entering 4 * "Directory

Number" followed by SEND.

34.5.5.8 Completion of calls to busy subscribers

In a situation where a calling party A encounters busy of congestion on the B side, the network may offer the possibility to apply the CCBS supplementary service. If subscriber A, after being notified that CCBS is possible and during the period where the retention timer is running (minimum 15 seconds), enters 5 followed by SEND, this shall be interpreted as CCBS activation. Entering of 5 SEND by subscriber A in any other situation as described above shall not be interpreted as CCBS activation.

34.5.6 Other handling of supplementary services

34.5.6.1 Multiple Subscriber Profile

34.5.6.1.1 Registering an alternative profile

An alternative profile is registered by entering the profile ID of the new profile, as illustrated:

* 59n # SEND

Where n is the identity of the profile desired.

An indication is given to the user showing whether this procedure was successful.
The ID of the registered profile and other provisioned profiles may be determined by interrogation on entering *#59# SEND
The profile so registered shall be used for all further ~~MS~~UE originated activities and CISS operations unless another profile is selected, or an alternative profile is registered.

34.5.6.1.2 Selecting an alternative profile on a per call basis

An alternative profile to the registered profile is selected by entering the profile ID of the new profile along with the desired Directory Number, as illustrated:

DN *59n# SEND

Where n is the identity of the profile desired

Continued processing of the call shall implicitly indicate that the selection was successful, there shall be no explicit indication given to the user concerning successful execution this selection procedure.
It is assumed that *# (i.e. interrogation) will not apply to call set-up.

34.5.6.2 Calling Line Identification Presentation (CLIP)

The CLIP Supplementary Service is defined in GSM 02.81~~[22]~~[12]

34.6.6.2.1 Presentation of Information

If CLIP has been provisioned for the subscriber and the ~~MS~~UE is capable of displaying the line identification then for each MT call the ~~MS~~UE should either:

display the calling line identity or;
display the reason why the line identity is not available as indicated by the Presentation Indicator

34.6 SIM/USIM interfaces

34.6.1 Entry of PIN and PIN2

After insertion of the IC card while the ~~MS~~UE is switched on, or when the ~~MS~~UE is switched on while the IC card is inserted, or when the ~~MS~~UE is switched on in the case of a plug-in SIM, an indication is given to the user that the PIN must be entered, unless the PIN is not applicable.
If the user wishes to perform a function protected by PIN2, an indication shall be given to the user that PIN2 must be entered.
The PIN or PIN2 being entered is not revealed in any way. The PIN or PIN2 check is performed by entering the # function.

34.6.2 Change of PIN or PIN2

The following procedure permits the user to change the PIN or PIN2 in the SIM/USIM:

PIN: **04*OLD_PIN*NEW_PIN*NEW_PIN#

PIN2: **042*OLD-PIN2*NEW_PIN2*NEW_PIN2#

Note that the SEND function is not used in these procedures.
An indication is given to the user showing whether this procedure was successful.

34.6.3 Unblocking of PIN or PIN2

The following procedure permits the user to unblock the PIN or PIN2:

PIN: **05*PIN_UNBLOCKING_KEY*NEW_PIN*NEW_PIN#

PIN2: **052*PIN2_UNBLOCKING_KEY*NEW_PIN2*NEW_PIN2#

Note that the SEND function is not used in these procedures.
The new PIN or PIN2 must be entered whether or not it is intended to change the PIN or PIN2. An indication is given to the user showing whether this procedure was successful.

34.6.4 Reading the abbreviated dialling code

An abbreviated dialling code shall be able to be read using the following procedure:

N(N)(N)#

Alternative additional procedures are also permitted.

34.6.5 Status information - return codes

The SIM gives status information, as responses to instructions~~, in two-byte codes (see GSM 11.11 [21] clause "Status Conditions Returned by the Card")~~. Some of the possible return codes are deeply related to the user's actions and should therefore be indicated to her.
It is mandatory to give the user the appropriate indication (respectively) when the following codes appear:

code		description;
92 40	-	Memory Problem (eg. Update impossible);
98 04	-	Access conditions not fulfilled (eg. secret code verify rejected);
98 40	-	Unsuccessful CHV verification, no attempt left (eg. Secret code locked);
6F XX	-	Technical problem with no diagnostic given. However, if this code is returned by the SIM in response to an ENVELOPE (SMS-PP DOWNLOAD) or ENVELOPE (CELL BROADCAST DOWNLOAD), then no indication shall be given to the user, since in this case the code is not related to a user action.

The status information indication can be a dedicated lamp, text-string or others, as long as it is unambiguously made available to the user via the MMI.
As regards all other codes, it is left to the manufacturers' discretion whether and how the user shall be informed.

34.7 Presentation of IMEI

The following procedure shall instruct the ME to display its IMEI:

*#06#

The procedure shall be accepted and performed with and without an inserted SIM. The ME shall then display the 14 digits of the IMEI (not including the spare digit), the Check Digit and optionally the Software Version Number as defined in TS 22.016 ~~[6]~~[4] (as a single string, in that order).

~~5 The basic public MMI~~

~~5.1 General~~

~~In order to improve the standardization of the MMI for Mobile Stations intended for general use by the public to access~~

~~voice services, the following additional specification is provided. Equipment which meets this specification may quote "Approved to 22.030 Section 5" in its specification.~~

~~This procedure is intended for Mobile Stations used by unfamiliar users, where instructions will be limited, for example in fleet cars, hire cars and payphones (cash, credit card, smart card, prepaid card, etc...).~~

~~The organization providing the facility may require "Approved to 22.030 Section 5" as part of its procurement specification.~~

~~The use of this clause 5 of the specification is not mandatory.~~

~~Use of "Approved to 22.030 section 5" is restricted to Mobile Stations which pass Type Approval testing in respect to this clause.~~

~~A manufacturer who wishes his equipment to be tested to this section for approval must declare his requirement on submission.~~

~~This specification covers the basic features of call origination and call termination. It specifies features which are mandatory for the Basic Public MMI. These are additional to the other clauses of TS 22.030 which still apply and provision of additional features and facilities is not precluded unless otherwise stated.~~

~~Guidelines for the application and design of pictographic instructions and the use of the symbol for telephone are to be found in ITU-T Recommendation E.121 [25] (Red Book, Vol. II - Fasc. II.2).~~

~~5.2 Basic public MMI specific features~~

~~5.2.1 Keyboard layout~~

- ~~Layout of 12-keypad as per figure 5.2.~~
- ~~Layout of all other keys are optional.~~
- ~~Control key functions: No additional functions standardized (exception: see subclause 5.2.3).~~
- ~~Control key symbols: No additional standardization.~~
- ~~Control key positions: No requirements.~~

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

~~Figure 5.2~~

~~5.2.2 Number entry~~

- ~~No restriction on number entry or editing.~~

~~5.2.3 Call control~~

- ~~A hand-set shall be present to place and receive calls.~~
- ~~SEND and END function keys are mandatory for execution of call initiation/termination respectively.~~

~~5.2.4 Call acceptance~~

- ~~On receiving "Ring Alert", the user may lift the hand-set "Off Hook" or press the SEND function key.~~

~~5.2.5 Call initiation~~

~~5.2.5.1 "Off Hook" call initiation~~

1) ~~Lift the hand-set "Off Hook" - Dial tone is presented.~~

2) ~~Enter number. Dial tone is cancelled after entry of the first digit (including * and #).~~

3) ~~Press SEND-function key.~~

4) ~~If the unit is "Off Hook" and the SEND-function key is not pressed, call set up may be automatically initiated after expiry of a time-out of 5 seconds. The time-out shall be restarted after every digit entry.~~

5) ~~The call initiation is stopped by replacing the hand-set "On Hook".~~

~~5.2.5.2 "On Hook" call initiation~~

~~Time-out dialling in "On Hook" - mode is not allowed for mobiles fitted with an "On Hook" dialling feature.~~

~~5.2.6 Call termination~~

- ~~The call is terminated by replacing the hand-set "On Hook" or by pressing the END-function key.~~

- ~~The call may also be terminated by e.g. replacing "On Hook" by B-party, radio path interruption and invalidation of payment.~~

~~5.2.7 Supplementary services control~~

~~The primary function of the * and # on the Basic Public MMI will be for control of supplementary services in accordance with the procedures defined in subclause 4.5. of this specification.~~

~~5.2.8 Payment~~

~~If the MS requires to be set up with some means of payment e.g. cash, prepaid card or credit card, a "payment indication" will be presented to the user when payment is required at the initial step (see subclause 5.2.5, Call Initiation Steps).~~
~~When a call is in progress and payment which has been made is nearly used up, the payment indication shall be presented again inviting the user to make further payment.~~
~~When sufficient payment has been entered the payment indication shall be switched off.~~
~~The requirement for payment prior to the origination of an emergency call is not precluded.~~

~~5.2.9 Country/PLMN selection~~

~~PLMN selection shall be in accordance with TS 22.011 [4], but "automatic" shall be the default mode.~~

Annex A (normative): ~~MS~~UE access mealy graph

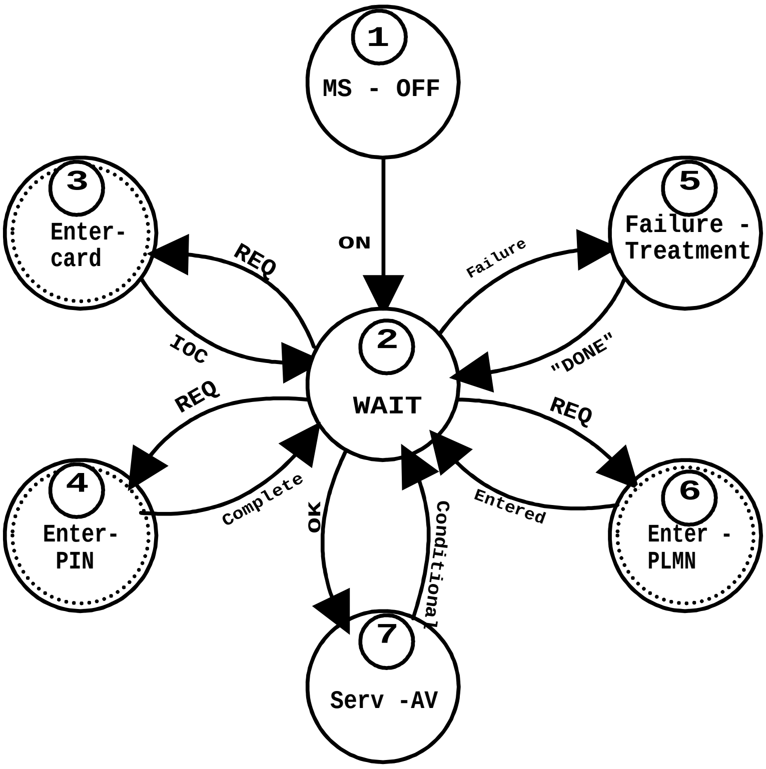


Figure A.1: Mealy-graph for the ms access procedure

Assumptions and requirements:

1) Emergency calls shall be possible in all states, except in state 1.

2) Power-off should cause transition to state 1 from all other states.

3) The actions to be taken in state 5 is not defined.

~~4) Basic requirements concerning indications and procedures for the different states are given in specifications GSM 02.07 [3], TS 22.011 [4], 02.17 [5] and 02.40 [7].~~

~~5) Additional indications to those in item 4 above may be given in all states and at all events.~~

4~~6~~) Realization of the dotted states (3, 4 and 6) depends upon the network requirements and the type of ~~MS~~UE.

Description of the states of the ~~MS~~UE access procedure

1) ~~MS~~UE-OFF: The ~~MS~~UE is in OFF-condition. This means that the equipment is not active as an ~~MS~~UE in a PLMN.

2) WAIT: Waiting for the completion of the ~~MS~~UE access conditions, which are related to the type of ~~MS~~UE and to the PLMN, where in the ~~MS~~UE is roaming (e.g. location updating).

3) ENTER CARD: Request for entering of the subscriber card, (e.g. when no built in SIM module is available).

4) ENTER PIN: Request for entering of the correct PIN.

5) FAILURE TREATMENT: Waiting for removal the actual failure condition.

6) ENTER PLMN: Request for selection of PLMN.

7) SERV-AV: The ~~MS~~UE is in a ready state. PLMN services are available to the user.

Description of the transitions between ~~MS~~UE access states

ON: The equipment becomes active as an ~~MS~~UE in a PLMN.

REQ: A request for user activity.

IOC: Insertion of a subscriber card with SIM-module.

COMPLETE: The PIN has been entered.

ENTERED: A PLMN choice has been done.

FAILURE: A failure condition has occurred in any other state during the ~~MS~~UE access procedures.

CONDITIONAL: One of the conditions the ~~MS~~UE is waiting for in WAIT state has been lost. The ~~MS~~UE goes back to the WAIT state.

"DONE": The ~~MS~~UE access failure condition has been corrected.

OK: All the conditions the ~~MS~~UE is waiting for in the WAIT state are accomplished.

Annex B (normative): Codes for defined Supplementary Services

Table B.1: Input information for handling of defined Supplementary Services

	Supplementary Service	Service Code	SIA	SIB	SIC
22.067					
	eMLPP	75 and 75n		where n=0-4	
22.072					
	CD	66			
22.081					
	CLIP	30	-	-	-
	CLIR	31	-	-	-
	COLP	76	-	-	-
	COLR	77	-	-	-
In temporary mode, to suppress CLIR for a single call, enter:	" * 31 # <called number> SEND "				
In temporary mode, to invoke CLIR for a single call enter:	" # 31 # <called number> SEND "				
22.082					
	CFU	21	DN	BS	-
	CF Busy	67	DN	BS	-
	CF No Reply	61	DN	BS	T
	CF Not Reachable	62	DN	BS	-
all CF		002	DN	BS	T
all conditional CF		004	DN	BS	T
22.083					
	WAIT	43	BS	-	-
		see section 4.5.5			
	HOLD	see section 4.5.5			
22.084					
	MPTY	see section 4.5.5			
22.087					
	UUS Service 1	361	R	-	-
	UUS Service 2	362	R	-	-
	UUS Service 3	363	R	-	-
	all UUS Services	360	R	-	-
	If UUS shall be activated when originating a call, enter: " * 36X * R # <called number> SEND" (X is indicating the requested UUS service)				

Table B.1(concluded): Input information for handling of defined Supplementary Services

	Supplementary Service	Service Code	SIA	SIB	SIC
22.088					
	BAOC	33	PW	BS	-
	BAOIC	331	PW	BS	-
	BAOIC exc home	332	PW	BS	-
	BAIC	35	PW	BS	-
	BAIC roaming	351	PW	BS	-
all Barring Serv.		330	PW	BS	-
Outg. Barr. Serv.		333	PW	BS	
Inc. Barr. Serv.		353	PW	BS	
22.091					
	ECT	96		see section 4.5.5	
	22.093				
	CCBS	37	n where n=1-5	See Section 4.5.5	
	22.096				
	CNAP	**300**	**-**	**-**	**-**
	22.097				
	MSP	59n	PW	where n=1-4	

DN = Directory Number;
PW = Password (see subclause 4.5.4);
BS = Basic Service Group (if required) - see annex C;
T = No Reply Condition Timer (5-30 seconds);
R = UUS required option.

SI required Y = Yes;
N = No;
- = Not applicable.

"UUS required" option

For the "UUS required" option two values are defined:

R = 0 UUS not required;
R = 1 UUS required.

NOTE: If the "UUS required" option is requested for a call, the call will only be established if the requested UUS capabilities are available.

If the "UUS required" option is not contained in an activation request UUS shall be activated without the UUS required option.

Annex C (normative): Codes for Tele- and bearer services

Tele- and Bearer Service Supplementary Information codes (SIb).
Alternate and speech/data services are included with the equivalent data service.

Basic Service group number (note)	Telecommunication Service	MMI Service Code
1 to 12	All tele and bearer services	no code required
	Teleservices	
1 to 6, 12	All teleservices	10
1	Telephony	11
2 to 6	All data teleservices	12
6	Facsimile services	13
2	Short Message Services	16
1, 3 to 6, 12	All teleservices except SMS	19
12	Voice group services	
	Voice Group Call Service (VGCS)	17
	Voice Broadcast Service (VBS)	18
	Bearer Service	
7 to 11	All bearer services	20
7	All async services	21
8	All sync services	22
8	All data circuit sync	24
7	All data circuit async	25
13	All GPRS bearer services	99

NOTE: See TS 22.004 [2] for definition of Basic Service groups.

NOTE: "All GPRS bearer services" are not included in "All tele and bearer services" and "All bearer services".

The grouping implies that if e.g. code 25 is used, the Supplementary Service procedure concerned applies to all Asynchronous Data Circuit mode Bearer Services subscribed to.

Tele-and Bearer Service Supplementary Information Codes (SIb) for services not defined by GSM

Code as defined in TS 29.002 ~~[19]~~[11]	Telecommunication Service	MMI Service Code
PLMN specific teleservices:		
11010000	All PLMN specific teleservices	50
11010001	PLMN specific teleservice 1	51
11010010	PLMN specific teleservice 2	52
11010011	PLMN specific teleservice 3	53
11010100	PLMN specific teleservice 4	54
11010101	PLMN specific teleservice 5	55
11010110	PLMN specific teleservice 6	56
11010111	PLMN specific teleservice 7	57
11011000	PLMN specific teleservice 8	58
11011001	PLMN specific teleservice 9	59
11011010	PLMN specific teleservice 10	60
11011011	PLMN specific teleservice 11	61
11011100	PLMN specific teleservice 12	62
11011101	PLMN specific teleservice 13	63
11011110	PLMN specific teleservice 14	64
11011111	PLMN specific teleservice 15	65
PLMN specific bearer services:		
11010000	All PLMN specific bearer services	70
11010001	PLMN specific bearer service 1	71
11010010	PLMN specific bearer service 2	72
11010011	PLMN specific bearer service 3	73
11010100	PLMN specific bearer service 4	74
11010101	PLMN specific bearer service 5	75
11010110	PLMN specific bearer service 6	76
11010111	PLMN specific bearer service 7	77
11011000	PLMN specific bearer service 8	78
11011001	PLMN specific bearer service 9	79
11011010	PLMN specific bearer service 10	80
11011011	PLMN specific bearer service 11	81
11011100	PLMN specific bearer service 12	82
11011101	PLMN specific bearer service 13	83
11011110	PLMN specific bearer service 14	84
11011111	PLMN specific bearer service 15	85

3G CHANGE REQUEST

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

22.081 CR 002 Current Version: 3.0.1

3G specification number ↑ *↑ CR number as allocated by 3G support team*

For submision to TSG SA1	for approval	X	*(only one box should*
list TSG meeting no. here ↑	for information		*be marked with an X)*

Form: 3G CR cover sheet, version 1.0 *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/3GCRF-xx.rtf*

Proposed change affects: USIM [ ] ME [ ] UTRAN [ ] Core Network [ ]
(at least one should be marked with an X)

Source: TSG SA1 **Date:** 25/11/1999

Subject: Editorial update to TS 22.081 in order to include 3G systems

3G Work item: 3TS/SA-0122081

Category: *(only one category shall be marked with an X)*

F	Correction	
A	Corresponds to a correction in a 2G specification	
B	Addition of feature	
C	Functional modification of feature	
D	Editorial modification	X

Reason for change: Text referring to the GSM system needs to be changed to refer to both the GSM and 3G systems.

Clauses affected: 3.2.1, 3.7

Other specs affected:

Other 3G core specifications		→ List of CRs:
Other 2G core specifications		→ List of CRs:
MS test specifications		→ List of CRs:
BSS test specifications		→ List of CRs:
O&M specifications		→ List of CRs:

Other comments:

help.doc

<--------- double-click here for help and instructions on how to create a CR.

3G TS 22.081 V3.0.1 (1999-10)

Technical Specification

3rd Generation Partnership Project;
Technical Specification Group Services and System Aspects;
Line identification Supplementary Services;
Stage 1
(3G TS 22.081 version 3.0.1)

3 Connected Line Identification Presentation (COLP)

3.1 Definition

The Connected Line Identification Presentation (COLP) Supplementary Service provides the calling party with the possibility to receive the line identity of the connected party.

3.2 Description

3.2.1 Description

This Supplementary Service is not a dialling check but an indication to the calling subscriber of the connected line identity in a full ISDN/~~GSM~~ PLMN environment, the connected line identity shall include all the information necessary to unambiguously identify the connected party.
The network shall deliver the connected line identity to the calling party regardless of the terminal capability to handle the information.

……..

3.7 Interworking considerations

According to national network specific rules the COLP Supplementary Service need not be applicable, if at least one of the two parties is not an ISDN or ~~GSM~~ PLMN subscriber.

TSG-SA Working Group 1 meeting #6 **TSG S1** ***(99)999***
San Diego, USA, 29 Nov-3 Dec 1999 Agenda: 6

3G CHANGE REQUEST

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

22.085 CR 002 Current Version: 3.0.1

3G specification number ↑ *↑ CR number as allocated by 3G support team*

For submision to TSG	SA1	for approval	X	*(only one box should*
list TSG meeting no. here ↑		for information		*be marked with an X)*

Proposed change affects: USIM [ ] ME [ ] UTRAN [ ] Core Network [ ]
(at least one should be marked with an X)

Source: TSG SA1 **Date:** 25/11/1999

Subject: Editorial update for GSM/3GPP use.

3G Work item: 3TS/SA-0122085

Category: *(only one category shall be marked with an X)*

F	Correction	
A	Corresponds to a correction in a 2G specification	
B	Addition of feature	
C	Functional modification of feature	
D	Editorial modification	X

Reason for change: Applicability of Supplementary Services to telecommunication services is shown in TS 22.004. A reference to 22.004 is added to section “Applicability to telecommunication services” in TS 22.085 instead of updating the existing text to clarify that CUG is applicable to “connection oriented” telecommunication services.

Clauses affected: 1.2.2

Other specs affected:

Other 3G core specifications		→ List of CRs:	
Other 2G core specifications		→ List of CRs:	
MS test specifications		→ List of CRs:	
BSS test specifications		→ List of CRs:	
O&M specifications		→ List of CRs:	

Other comments:

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<--------- double-click here for help and instructions on how to create a CR.

3G TS 22.085 V3.0.1 (1999-10)

Technical Specification

3rd Generation Partnership Project;
Technical Specification Group Services and System Aspects;
Closed User Group (CUG) Supplementary Services - Stage 1
(3G TS 22.085 version 3.0.1)

1.2.2 Applicability to telecommunication services

The applicability of this Supplementary Service is defined in GSM 22.004 [2].

~~The CUG Supplementary Service is applicable to all telecommunication services, except Emergency calls, SMS, dedicated PAD access and dedicated Packet access~~.

3G CHANGE REQUEST

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

22.101 CR 026 Current Version: 3.7.0

3G specification number ↑ *↑ CR number as allocated by 3G support team*

For submision to TSG SA#6 for approval X *(only one box should be marked with an X)*
list TSG meeting no. here ↑ for information

Form: 3G CR cover sheet, version 1.0 *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/3GCRF-xx.rtf*

Proposed change affects: USIM X ME X UTRAN X Core Network X
(at least one should be marked with an X)

Source: SA WG1 **Date:**

Subject: Mainly editorial update for GSM/3GPP use.

3G Work item:

Category:
(only one category shall be marked with an X)

F	Correction	
A	Corresponds to a correction in a 2G specification	
B	Addition of feature	
C	Functional modification of feature	X
D	Editorial modification	

Reason for change: The transfer of GSM specifications for 3GPP has created need to clarify requirements specifically related PS and CS domain. This CR updates 22.101 to be applicable to both the GSM and 3G systems. Further new annex is included covering big parts of GSM 02.07 including new UE requirements.

Clauses affected: All clauses

Other specs affected:

Other 3G core specifications		→ List of CRs:
Other 2G core specifications		→ List of CRs:
MS test specifications		→ List of CRs:
BSS test specifications		→ List of CRs:
O&M specifications		→ List of CRs:

Other comments:

help.doc

<--------- double-click here for help and instructions on how to create a CR.

1 Scope

This Technical Specification (TS) describes the Service Principles for ~~of~~ ~~the Universal Mobile Telecommunications System (UMTS)~~PLMNs specified by 3GPP.

~~The European initiative to develop UMTS should be seen as part of the policy to provide more advanced capabilities than can be anticipated for pre-UMTS systems.~~ 3GPP specifications ~~UMTS~~ provide~~s~~ integrated personal communications services. ~~UMTS PLMN operates in parallel with pre-UMTSexisting technologies (e.g. GSM, DCS 1800, DECT, TETRA etc.) which must be allowed to achieve their full potential.~~ ~~UMTS is a~~The system ~~that~~ 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.

~~UMTS~~ 3GPP specifications allow~~is~~ 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.

~~UMTS~~ 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 2 Mbits/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;

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.
- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.

2.1 Normative References

~~[1] UMTS 22.25 "Universal Mobile Telecommunications System (UMTS): Quality of Service and Network Performance"~~

[1~~2~~] ~~UMTS~~ TS 22.105 "~~Universal Mobile Telecommunications System (UMTS):~~ Services and Service Capabilities~~ related to Service Usage~~"

[2~~3~~] TS ~~UMTS 22.XX~~ 22.121: "~~Universal Mobile Telecommunications System (UMTS):~~ Virtual Home Environment (VHE), Stage 1"

[3~~4~~] TS~~UMTS~~ 22.038: "SIM application toolkit, stage 1"

~~2.2 Informative references~~

~~[1] ITU-T Draft Recommendation F.700: "Framework Recommendation for audio-visual/multimedia services";~~

~~[2] ITU-T Draft Recommendation F.SFEA: "Service Features and Operational Provisions in IMT-2000".~~

[4~~3~~] TS ~~GSM 0~~22.001: "~~Digital cellular telecommunications system (Phase 2+);~~ Principles of Circuit telecommunication services supported by a ~~GSM~~ Public Land Mobile Network (PLMN)".

[5~~4~~] TS ~~GSM 0~~22.004: ~~"Digital cellular telecommunications system (Phase 2+);~~ General on supplementary services"

[6~~5~~] TS ~~GSM 0~~22.030: "~~Digital cellular telecommunications system (Phase 2+);~~ Man-Machine Interface (MMI) of the User Equipment ~~Mobile Station~~ (UE~~MS~~)"

[7~~6~~] ~~GSM 0~~TS 22.066: "~~Digital cellular telecommunications system (Phase 2+);~~ Support of Mobile Number Portability (MNP); Service description; Stage 1"

[8~~7~~] ~~GSM 0~~TS 22.079: "~~Digital cellular telecommunications system (Phase 2+);~~ Support of Optimal Routing; Stage 1"

[9~~8~~] TS 22.129: "Handover Requirements between UMTS and GSM or other Radio Systems"

[~~9~~10] TS 33.102: "Security Architecture"

[11~~10~~] TS 22.011: "Service Accessibility"

[12~~11~~] TS 22.016: "International mobile Station Equipment Identities (IMEI)"

[13] GSM 04.08: "Digital cellular telecommunications system (Phase 2+); Mobile Radio Interface Layer 3 Specification"

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

[15] TS 21.133: "Security Threats and Requirements"

[16] TS 33.120: "Security Principles"

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

[18] GSM 02.09: "Digital cellular telecommunications system (Phase 2+); Security Aspects"

[19] TS 31.102: "USIM Application Characteristics"

[20] TS 22.121: "Architectural Requirements for Release 99"

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of this TS, the following definitions apply:

Authentication: a property by which the correct identity of an entity or party is established with a required assurance. The party being authenticated could be a user, subscriber, home environment or serving network.

Bearer: a bearer capability of defined capacity, delay and bit error rate, etc.

Bearer capability: a transmission function which the ~~mobile station~~user equipment requests to the network.

~~**Cipher key:** a code used in conjunction with a security algorithm to encode and decode user and/or signalling data.~~

Confidentiality: the avoidance of disclosure of information without the permission of its owner.

Home Environment: the home environment is responsible for enabling a user to obtain ~~UMTS~~ services in a consistent manner regardless of the user's location or terminal used (within the limitations of the serving network and current terminal).

IC Card: a card holding an Integrated Circuit containing subscriber, end user, authentication and/or application data for one or more applications.

Integrity: (in the context of security) is the avoidance of unauthorised modification of information.

~~**International mobile user number (IMUN):** The International Mobile User Number is a diallable number allocated to a UMTS user.~~

Label: A number or name as defined below.

Mobility: the ability for the user to communicate whilst moving independent of location.

Multimedia service: Multimedia services are services that handle several types of media such as audio and video in a synchronised way from the user's point of view. A multimedia service may involve multiple parties, multiple connections, and the addition or deletion of resources and users within a single communication session.

Name: A name is an alpha numeric label used for identification of end users and may be portable.

Number: A string of decimal digits that uniquely indicates the public network termination point. The number contains the information necessary to route the call to this termination point.

A number can be in a format determined nationally or in an international format. The international format is known as the International Public Telecommunication Number which includes the country code and subsequent digits, but not the international prefix.

Number portability: where the provision of directory~~diallable~~ numbers is independent of home environment and/or serving network.

One Stop Billing: one bill for all charges incurred using ~~UMTS~~PLMN services.

Quality of Service: the collective effect of service performances which determine the degree of satisfaction of a user of a service. It is characterised by the combined aspects of performance factors applicable to all services, such as:

- service operability performance;
- service accessibility performance;
- service retention~~retainability~~ performance;
- service integrity performance;
- and other factors specific to each service.

Roaming: the ability for a user to function in a serving network.

Security: the ability to prevent fraud as well as the protection of information availability, integrity and confidentiality.

Service: is set of functions offered to a user by an organisation.

Service Control: is the ability of the user, home environment or serving environment to determine what a particular service does, for a specific invocation of that service, within the limitations of that service.

Serving Network: the serving network provides the user with access to the services of home environment.

Subscriber: the responsibility for payment of charges incurred by one or more users may be undertaken by another entity designated as a subscriber. This division between use of and payment for services has no impact on standardisation.

Supplementary service: is a service which modifies or supplements a basic telecommunication service. Consequently, it cannot be offered to a customer as a standalone service. It must be offered together with or in association with a basic telecommunication service. The same supplementary service may be common to a number of telecommunication services.

Teleservice: is a type of telecommunication service that provides the complete capability, including terminal equipment functions, for communication between users according to standardised protocols and transmission capabilities established by agreement between operators.

User: is a logical, identifiable entity which uses ~~UMTS~~ services.
User Profile: is the set of information necessary to provide a user with a consistent, personalised service environment, irrespective of the user's location or the terminal used (within the limitations of the terminal and the serving network).
User Equipment: is a combination of mobile equipment (ME) and SIM/USIM.
USIM: User Service Identity Module is an application residing on the IC-Card used for accessing ~~UMTS~~ services with appropriate security.
Virtual Home Environment: the virtual home environment is a system concept for personalised service portability between serving networks and between terminals.

3.2 Abbreviations

For the purposes of this TS, the following abbreviations apply:

BER	Bit Error Rate
~~B-ISDN~~	~~Broadband ISDN~~
~~DAM~~	~~DECT Authentication Module~~
~~DECT~~	~~Digital Enhanced Cordless Telecommunications~~
DTMF	Dual Tone Multiple Frequency
~~ECTRA~~	~~European Committee of Telecommunications Regulatory Affairs~~
~~ETNS~~	~~European Telecommunications Numbering Space~~
ETSI	European Telecommunications Standards Institute
FDD	Frequency Division Duplex
GSM	Global System for Mobile Communications
~~HF~~	~~Human Factors~~
~~IEC~~	~~International Electrotechnical Commission~~
IMT-2000	International Mobile Telecommunications 2000
~~IMUN~~	~~International Mobile User Number~~
IN	Intelligent Network
ISDN	Integrated Services Digital Network
~~ISO~~	~~International Organisation for Standardisation~~
ITU	International Telecommunication Union
LAN	Local Area Network
ME	Mobile Equipment
MMI	Man Machine Interface
MO	Mobile Origination
~~MS~~	~~Mobile Station~~
UE	User Equipment
MT	Mobile Termination
O&M	Operations and Maintenance
PBX	Private Branch eXchange
PC	Personal Computer
PCMCIA	Personal Computer Memory Card International Association
PIN	Personal Identity Number
PNP	Private Numbering Plan
POTS	Plain Old Telephony Service
QoS	Quality of Service
SIM	Subscriber Identity Module
SMS	Short Message Service
TDD	Time Division Duplex
~~TE9~~	~~Terminal Equipment 9 (ETSI sub-technical committee)~~
UICC	UMTS IC Card
USIM	User Service Identity Module
UMTS	Universal Mobile Telecommunications System
VHE	Virtual Home Environment

4 General

4.1 Aims of 3GPP specifications~~UMTS~~

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 ~~pre-UMTS~~ existing systems, delivering mass market low-cost digital telecommunication services.
~~UMTS therefore seeks~~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

~~Pre-UMTS~~Existing systems have largely standardised the complete sets of teleservices, applications and supplementary services which they provide. As a consequence, substantial re-engineering is often required to enable new services to be provided and the market for services is largely determined by operators and standardisation. This makes it more difficult for operators to differentiate their services.
~~UMTS~~ 3GPP shall therefore standardise service capabilities and not the services themselves. 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. Section 6 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, other teleservices, user applications and supplementary services and enable the market for services to be determined by users and home environments.
~~The standardisation of service capabilities rather than the services themselves is a major differentiator between UMTS and pre-UMTS systems, see UMTS 22.xx.~~

4.3 Efficient Use of Network Resources

~~UMTS s~~Service capabilities shall take account of the discontinuous and asymmetric nature of most teleservices and user applications in order to make efficient use of network resources (particularly radio resources).
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 station~~mobile equipment relative speeds and traffic characteristics - see [2]). Although ~~UMTS~~ 3GPP aims to minimise the number of ~~UMTS~~ radio interfaces and to maximise commonality between them it~~UMTS~~ may utilise several radio interfaces, each optimised for different environments. Each radio interface might provide differing service capabilities. For ~~UMTS Phase1~~3GPP release 99, a single radio interface supporting two modes (TDD and FDD) is defined (UTRAN). The 3GPP Release 99 core network shall be capable of supporting the GERAN ~~BSS~~ as specified by ETSI SMG.
~~If more than one radio interface is defined for UMTS, the UMTS~~3GPP specifications ~~standard~~ shall provide a mechanism which will enable a user equipment (UE) ~~UMTS terminal~~ to adapt to different radio interfaces as necessary and to determine the service capabilities available. The ~~standard~~ specifications shall also provide a mechanism which will enable a ~~UMTS terminal~~UE to select radio interfaces capable of providing appropriate service capabilities.

4.4 Compatibility with Global Standards

~~UMTS~~ 3GPP specifications aims 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.
~~UMTS~~ 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 Virtual Home Environment

The above general principles plus the service architecture principles stated in section 6 specify all the capabilities of the virtual home environment (VHE).
~~UMTS~~ The 3GPP standards aim to provide the user with a comprehensive set of services and features, which have the "same look and feel" wherever they are used. For further information see ~~UMTS~~ 3GPP TS 22.121~~xxx~~ [2]. Especially the VHE shall provide for:

- a generic set of services / features and access capabilities, if the required service capabilities are available in the visited network;
- the means for serving network, home environments and user to re-use existing system capabilities to define their own specific features / services;
- user personalisation of features / services;
- a personalised service set being used via all ~~UMTS~~ access and transport networks, subject to physical limitations;
- the ability for the user to have access to personalised services from any suitable ~~UMTS terminal~~UE
- regional or network based variations, ~~/~~ enhancements to the basic services. ~~/ standard UMTS~~;
- future evolution of ~~UMTS~~ 3GPP specifications itself.

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 Domain and a Circuit Switched Domain. Networks and terminals may support only the PS domain, only the CS domain or both. For further information see TS 23.121 [20].

5 Principles for new service capabilities

5.1 General

The standard 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.

5.2 Multimedia

~~UMTS~~ 3GPP specifications shall support multimedia services and provide the necessary capabilities.
Multimedia services combine two or more media components (e.g. voice, audio, data, video, pictures) 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.
The 3GPP specifications~~UMTS~~ shall support single media services (e.g. telephony) and multimedia services(e.g. video telephony).All calls shall have potential to become multimedia calls and there shall be no need to signal, in advance, any requirement for any number of multimedia components. However, it shall be possible to reserve resources in advance to enable all required media components to be available.

5.3 Service Management Requirements

There will be increased demands for better customer care and cost reductions in managing mobile networks due to :

- the provision of sophisticated personal communications services;
- the expansion of the customer base beyond the business user base;
- the separation between entities of home environment and serving network; and
- drives for 'one stop' billing for a range of services.

In ~~pre-UMTS~~ existing mobile networks, Service Management has largely been concerned with the management of physical products (often from different vendors and having different network management interfaces). ~~UMTS~~ 3GPP specifications shall include ~~support~~ standardised protocols enabling network management of functionality rather than network management of products and enabling:

- the support of Virtual Home Environment;
- management of user profiles;
- support of number portability;
- control, creation and subscription of service capabilities and services;
- provision of 'one stop' billing;
- quality of service.

6 Service architecture

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

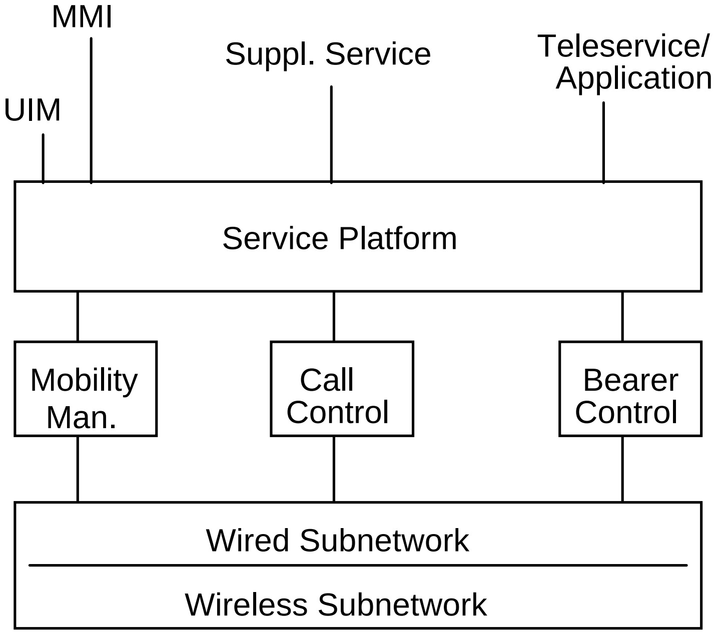


Figure 2: Service Architecture

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, and end user applications generally, via sub-networks 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 ~~provided by UMTS~~ 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/application in the fixed network and the bearers ~~provided by UMTS~~.

Adaptation/Interworking functions are required which take account of the discontinuous and/or asymmetrical nature of most teleservices/applications.

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

Supplementary service provision and control ~~provided by UMTS~~ 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 and/or end user applications to be supported at minimum cost, with minimum disruption of service and within the shortest possible time.

7 Quality of Service (QoS)

The ~~UMTS~~ Quality of Service (QoS) parameters should be identified together with appropriate parameter values which set targets to be reached when designing ~~UMTS standards~~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~~3~~] ~~and [4]~~.

8 Emergency Calls~~Security~~

~~This section covers service related security issues, more general security matters are considered in ETR 50901.~~

~~8.1 Security for the Serving Network and Home Environment~~

~~Charging information shall be incontestable and therefore require that the user be unambiguously identified, though there may be exceptions such as pre-paid cards. The UMTS IC Card (UICC) shall be physically present in order to make use of any services except for emergency calls.~~

~~The identification of the user should be based upon the UICC and steps should be taken to prevent fraudulent use of~~

~~stolen IC Card's. As far as possible the usage of stolen terminals should be prevented.~~
~~The standard shall cater for the ability for UMTS networks~~PLMNs ~~to authenticate each other.~~

~~8.2 Control of User Profiles and Supplementary Services~~

~~Although supplementary services may not be standardised the control of supplementary services will need to be implemented in a secure manner; e.g. call diversions to international numbers may be barred or limited in number at the discretion of the serving network.~~
~~The home environment shall have control of all aspects of user profiles.~~
~~Any changes to user profiles shall be done in a secure manner.~~

~~8.3 Security for the user~~

~~It should be possible for the user to authenticate the network when registering and before initiating a service if desired.~~
~~Steps shall be taken to ensure the privacy and integrity of sensitive information transferred between the user and all other entities; e.g. user identity and user traffic.~~

~~8.4 Emergency calls~~

PLMNs shall support an emergency call teleservice as defined in ~~GSM~~ TS 2~~0~~2.003 [14](TS12), which fulfils the following additional service requirements:
It shall be possible to establish an emergency speech call to the serving network. Emergency calls will be routed to the emergency services in accordance with national regulations ~~(GSM 02.03)~~. This may be based upon one or more default numbers stored in the ME ~~(3GPP TSGSM 022.030 [6])~~. It may also be possible to establish an emergency call without the need to dial a dedicated number, such as by use of a 'red button', or a linkage to a car air bag control. This functionality shall be available without a SIM/USIM being present. No other type of calls shall be accepted ~~by an ME~~ without a ~~U~~SIM/USIM.
The Emergency call teleservice is required only if the ~~MS~~UE supports telephony.

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

~~Further, the following requirement shall be fulfilled:~~
When a SIM/USIM is present, subscriber specific emergency call set-up MMI shall be provided. The operator shall specify preferred emergency call MMI(s) (e.g. 911 for US citizens or 110, 118 and 119 for Japanese citizens) for use in any (i.e. home or visited) PLMN. This 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: Release '98 and earlier SIM cards have the capability to store additional emergency call set-up MMI. However in many cases this has not been used.

When a SIM/USIM containing stored emergency numbers is present, only those numbers are identified as emergency numbers, i.e. default emergency numbers stored in the ME are ignored.
The following emergency numbers shall be stored in the ME: 000, 08, 112, 110, 911 and 999.

Note: Emergency numbers stored in the~~to~~ ME should not overlap with existing service numbers used by any operator.

It shall be possible for the serving network to obtain the number, which was used to initiate the emergency call. This will allow the network the option to route the call to different emergency call centres if appropriate. If the dialled digits are not recognised as an emergency service by the serving network, the call shall be routed to the default emergency service.

9 Numbering principles

The following section provides the requirements for numbering and identification of ~~UMTS~~ users:
~~(~~Note: L~~l~~abels are not required to be supported by the 3GPP release '99 specification~~standard~~.~~)~~

General requirements are listed in the following:

- The user shall be able to initiate communications with another party using a label / number to identify that party. This might be a logical label / number referring to a job function, and advertising response line etc. and would be resolved into a real terminal address by the ~~UMTS~~ system transparently to the user. Labels / numbers shall be capable of being stored in an address book which shall be accessible from any terminal that the user is registered on. Labels / numbers may be used to identify groups as well as individual terminals or people and shall allow extended character sets.

- 3rd party services should be reached by a label. Based on the selected charging policy for this services the calling party or/and the home environment of the calling party needs to be uniquely identified.

- Users also have requirements with regard to addressing for receipt of communications. The user shall be able to have a label / number of different persona (e.g. business and personal), each of which can be managed independently.

- When receiving communications, the recipient shall perceive the caller's label / number in the appropriate role. For example, when making a call as chairman of an 3GPP committee, then that persona will be presented as the caller ID. When making a personal call, then the underlying persona would be presented.

- In order to permit interworking with legacy networks, address interworking with common legacy network addressing shall be supported. In principle, this shall include interworking with any networking addressing scheme, but the following schemes listed below shall specifically be supported:

  - E.164,

  - E.168,

  - E.212,

  - X.121

  - Internet

9.1 ~~UMTS~~ Number portability

Some labelling / numbering schemes shall be fully independent of the supporting serving network and the home environment, allowing users to transfer this label to another home environment. For further information see ~~GSM~~ TS 2~~0~~2.066 [7].

An ~~International Mobile User Number (IMUN)~~MSISDN shall be allocated to each new user at the start of a ~~UMTS~~ 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 ~~UMTS~~ user shall be able to move subscription from one home environment to another without changing the ~~IMUN~~ 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 ~~IMUNs~~ MSISDNs from each of these domains as required.

9.2 Evolution path

Since 3GPP specifications~~UMTS~~ aim~~s~~ 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.

~~(For UMTS Phase 1) It is required that is shall be possible to identify UMTS users using GSM identities, namely IMSI, MSISDN and possibly TMSI and IMEI.~~

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

9.3 User / USIM Identification

It is a requirement that the user can be uniquely identified by the home environment from which the service is being obtained. This identification may be unknown to the serving network on which the user is roaming.

Serving networks need to be able to communicate with, authenticate and commercially deal with the home environment associated with any SIM/USIM being registered on their network. This shall require a SIM/USIM identity scheme which uniquely identifies each SIM/USIM, and a mapping scheme which allows the SIM/USIM identity to be used as a identifier with the "owning" home environment.

Serving networks also require to be able to route efficiently any communication to and from SIM/USIMs (or rather the devices on which they are registered). An address scheme is therefore required for operators to access and map any outgoing or incoming communication to SIM/USIMs and thus devices on their networks

9.4 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 TS 22.016 [12].

9.5 Home Environment / Serving Network Identification

Serving networks need to be able to communicate with, authenticate and commercially deal with the home environment associated with any SIM/USIM being registered on their network. This shall require a SIM/USIM identity scheme which uniquely identifies each SIM/USIM, and a mapping scheme which allows the SIM/USIM identity to be used as a identifier with the "owning" home environment.
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.

9.6 Service dependence / independence

Although a called party may be addressable via different means, he should be reachable independent of the medium. This would require a new functionality which can map label / number (digits) for call routing purposes. Networks might only support basic functionality while advanced databases might be offered by 3rd parties.
~~UMTS~~ The 3GPP specifications shall provide various methods to identify the service required, for example, via the number dialled or protocol headers.
It shall be possible for the home environment to change serving network(s) without changing ~~IMUN~~MSISDNs.
It shall be possible for several numbers to be associated with a single subscription on a single UICC.

9.7 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.
In addition, the user shall be able to choose the means to address the identity of a dialled number. For instance the number required to be dialled may be addressed by a spoken name.

NOTE: This may well be considered as a function of the equipment used to access the service and as such is not required to be standardised. However, the provision of such a facility needs to be provided across all terminal types used; fixed and mobile.

9.8 Numbering schemes

9.8.1 Multiple numbering scheme

The standards shall support the possiblity 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.

9.8.2 Single numbering scheme

The standards shall support the possiblity of allowing MT calls of different bearer types (eg 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.

9.9 Optimal routing

The implementation of the numbering scheme used ~~for UMTS~~ shall allow for optimal routing; i.e. routing shall not take place simply on the number dialled. See TS 2~~GSM 0~~2.079 [8] for some scenarios.

10 Human Factors and user procedures

As defined in the Service Provision Concepts subclause ~~of this ETS the UMTS system~~3GPP specifications should meet future communication requirements and shall be designed to be adaptable to provide new services as and when they are defined.

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 ~~of UMTS~~. 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 UMTS~~ of services should be as simple as possible with minimum input expected from the user;
- feedback, to the user from the various ~~UMTS~~ services, should be meaningful;
- any error recovery procedures provided should be simple to understand and execute.

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 ~~of UMTS~~ and therefore a common User Interface would be impractical.

Given the flexibility of the ~~UMTS~~ 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 supplementary services (~~GSM 02~~TS 22.004 [5]), may use MMI procedures specified in ~~GSM~~ TS 2~~0~~2.030 [6] and existing GSM MMI related ~~MS~~UE features (~~GSM 02.07~~Annex A) may also be used. In particular the following features are highly desirable for uniform UMTS UE implementation where appropriate:

- Mapping of numeric keys to European alphabetic keys to ensure compatible mnemonic dialing as defined in TS ~~0~~22.030 [6],
- "+" key function to enable one key international access as defined in Annex A~~02.07~~
- Structure of the MMI as described in ~~GSM 0~~TS 22.030 [6]
- Presentation of IMEI (International Mobile Equipment Identity) as defined in TS 2~~0~~2.030 [6]

11 UICC~~UMTS IC Card~~, USIM and Terminal

This clause defines the functional characteristics and requirements of the User Service Identity Module (USIM) ~~for use in UMTS~~. The USIM is an application residing on a UICC.

11.1 The USIM and User Profiles

11.1.1 The SIM/USIM

Every SIM/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 SIM/USIM.

The SIM/USIM shall be used to provide security features.

For access to ~~UMTS~~ services, provided via a ~~UMTS~~ home environment, a valid SIM/USIM shall be required.

The SIM/USIM shall support SIM Application Toolkit as specified in 3G TS 22.038 [3~~4~~].

The SIM/USIM shall reside on a UICC, 3GPP specifications~~UMTS~~ shall adopt both of the GSM SIM card physical formats. ~~New UMTS terminals may require o~~Other physical formats may also be supported. SIM/USIM specific information shall be protected against unauthorised access or alteration.

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

11.1.2 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. ~~IMUN~~MSISDN), 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.

11.1.3 UICC usage in 2G Terminals

It shall be possible to use the UICC in 2G terminals to provide access to GSM networks. In order to achieve that option, it shall be possible to store a module containing 2G access functionalities on the UICC which shall be accessed via the standard GSM SIM-terminal interface.

11.1.4 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. It shall be possible for the user to select the application 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. ~~when UMTS terminals and UICC are produced. Nevertheless, in the short term, it is safer to assume that only USIMs associated to one home environment will be stored on one UICC.~~

11.2 The UICC

~~Physical aspects of the UICC shall be defined by an appropriate committee. However there is a requirement to support a~~Access to services via GSM and 3G networks~~UMTS~~ with a single UICC shall be possible.

11.2.1 The~~ UMTS~~ UICC and Applications other than the USIM

It shall be possible for the UICC to host other applications in addition to the USIM, 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~~for UMTS~~ e.g. electronic commerce applications.
Examples of other UICC applications are: USIM, Phase 2+ SIM, 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.

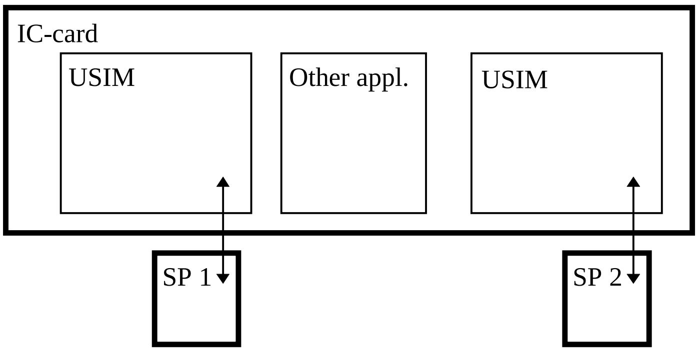


Figure 3 Example of a Multifunction UICC

11.3 Terminals and Multiple UICCs

A single terminal may support the use of multiple UICC (e.g with applications like USIM, SIM 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 or SIM, the requirements of 11.1.4 shall apply.
If the UICC with the active USIM or SIM is removed from the mobile terminal during a call (except for emergency

calls), the call shall be terminated immediately.

~~13 Service environment~~

~~The success of UMTS may depend upon its deployment in many regions of the world. Different regions of the world are likely to have widely different market needs for wireless based telecommunications, ranging from low cost provision of POTS (to users whose mobility is to be limited) to the provision of high bit rate and multimedia services (to highly mobile users).~~

~~The following scenarios should therefore be considered:~~

- ~~use of UMTS to primarily provide wideband services (up to 2Mbps).~~
- ~~use of UMTS to provide primarily telephony.~~
- ~~use of UMTS for narrow (up to 64kbps) and wideband services.~~
- ~~use of multi standard terminals to provide integrated services.~~

12~~4~~ Evolution

12~~4~~.1 Support of ~~pre UMTS~~existing services

The ~~UMTS standard~~3GPP specifications shall be capable of supporting ~~pre UMTS~~existing services in a manner which is transparent to the users of these services.

3GPP specifications~~UMTS~~ shall provide some mechanisms which permit 2G ~~pre UMTS~~ users to roam easily onto 3G networks~~UMTS~~ and access at least a minimum set ~~the~~of services. See Figure 4~~5~~ for clarification.

3GPP specifications~~UMTS~~ shall provide some mechanisms which permit 3G ~~UMTS~~ users to roam easily onto ~~pre-UMTS~~2G systems and access at least a minimum set ~~the~~of services.

12~~4~~.2 Provision and evolution of services ~~within UMTS~~

~~UMTS may be introduced before a complete set of UMTS standards is available. As one of the identified priority areas, UMTS service related standards need to be available at an early stage. If~~ Since a phased approach to the completion of specification~~standard~~s has~~is~~ been ad~~o~~apted ~~then~~ the same general service principals shall apply to each phase~~at an early stage~~.

~~UMTS n~~Networks shall be capable of providing a specified core set of capabilities. Responsibility for providing this core set of capabilities should lie with the serving network.

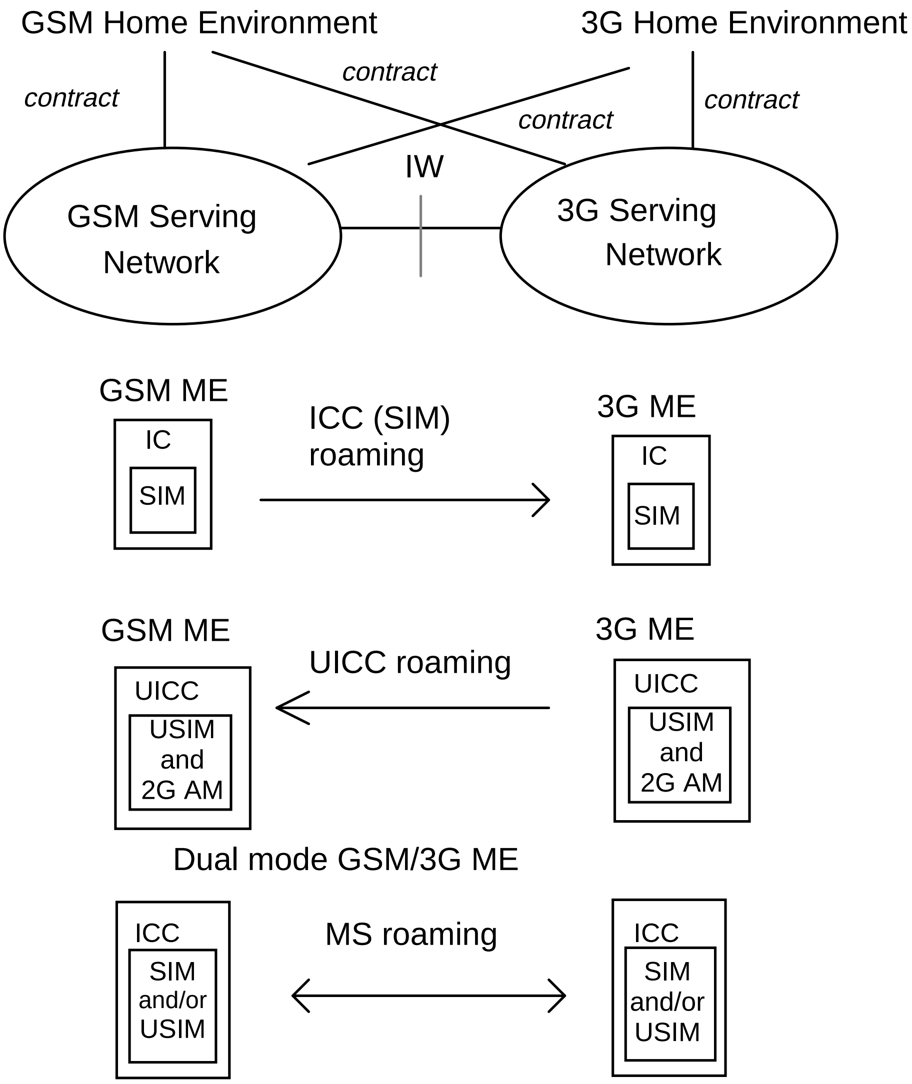
The core set of capabilities should permit ~~UMTS~~ home environment to offer a range of distinctive services including those which cannot be implemented on ~~pre-UMTS~~existing systems.

~~UMTS shall provide some mechanism which permits UMTS users to roam easily onto pre UMTS systems, with access to a minimum set of services.~~

It shall be possible for the home environment to develop services with full roaming capability. It should not be necessary for users to subscribe to more than one home environment in order to receive a particular service. For example a company may market an in car navigation/location system which uses UMTS as the core network. As far as users of the navigation service are concerned, that company is their home environment.

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

The standard shall provide a mechanism which allows a UMTS terminal to be easily upgraded so that it can access new services which are within the physical limitations of the terminal. Figure 4 shows as an example the support of roaming users between GSM and UMTS.



2G AM: 2G Access Module

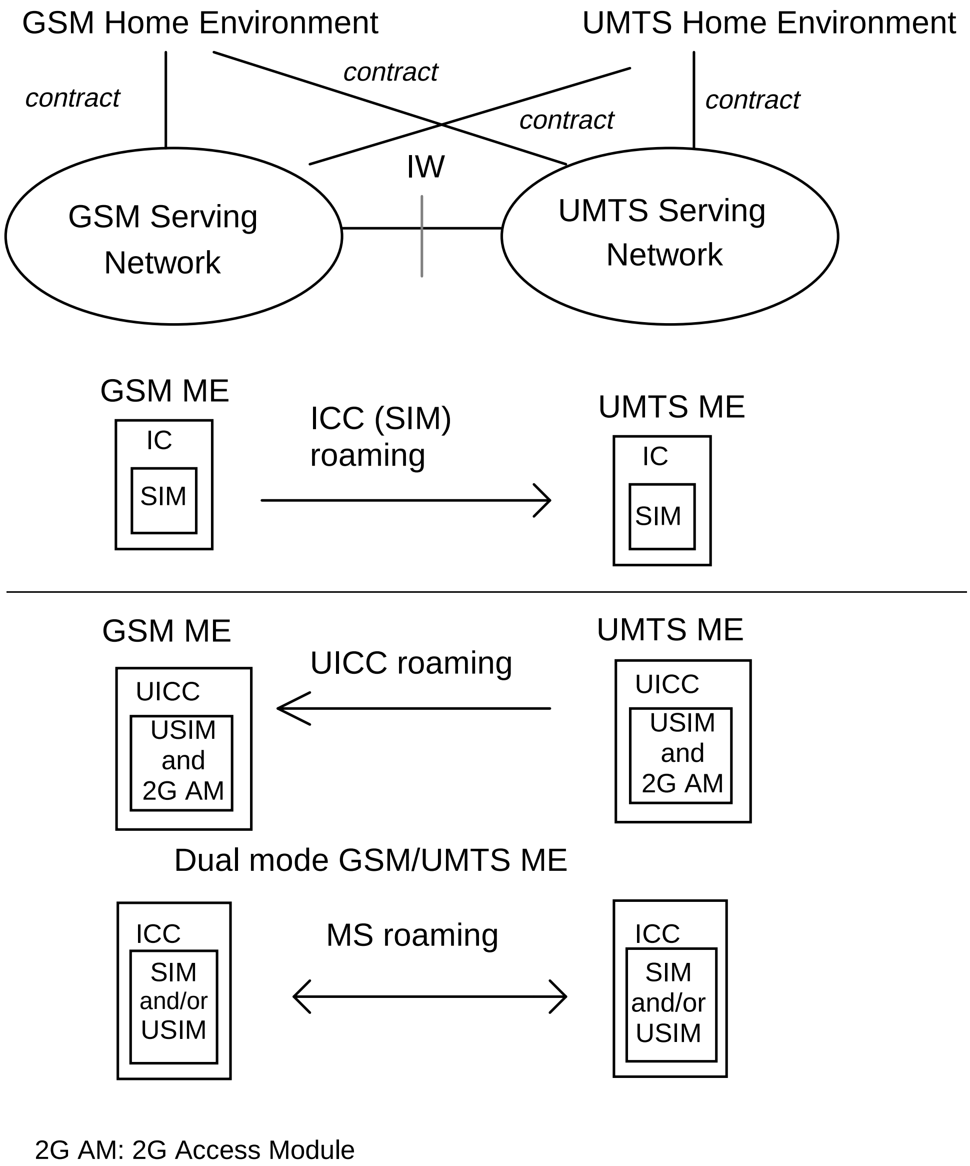


Figure 4 Roaming Users

13~~5~~ Types of features of ~~MS~~UEs

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

In order not to limit the possible types of ~~mobile station~~user sequipment they are not standardised~~ in UMTS. Anyhow some informative examples can be given to be the basis for mobile station discussions.~~ The ~~MS~~UE types could be categorised by their service capabilities rather than by their physical characteristics. Typical examples are speech only ~~MS~~UE, narrowband data ~~MS~~UE, wideband data ~~MS~~UE, data and speech ~~MS~~UE, etc.

In order to enhance functionality split and modularity inside the ~~mobile station~~user equipment the interfaces of ~~MS~~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.

~~MS~~UEs have to be capable of supporting a wide variety of teleservices and applications provided in ~~UMTS~~ PLMN environment. Limitations may exist on ~~MS~~UEs capability to support all possible teleservices and information types (speech, narrowband data, wideband data, video, etc.) and therefore functionality to indicate capabilities of an ~~MS~~UE shall be specified. ~~MS~~UEs should be capable of supporting new supplementary services without any changes in ~~MS~~UE.

The basic mandatory ~~MS~~UE requirements are:

- Encrypted terminal-UICC interface
- Support for GSM phase 2 and 2+ SIM cards, phase 1 5V SIM cards shall not be supported
- 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 ~~GSM 02.16~~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.
- Support for the execution of algorithms required for encryption;
- Support for the method of handling automatic calling repeat attempt restrictions as specified in ~~GSM 02.07~~TS 22.001 [4].
- At least one capability type shall be standardised for mobile terminals supporting the GERAN~~GSM BSS~~ and UTRAN radio interfaces.
- Under emergency situations, it may be desirable for the operator to prevent ~~MS~~UE users from making access attempts (including emergency call attempts) or responding to pages in specified areas of a ~~UMTS~~ network, see ~~GSM 02.11~~TS 22.011 [11].
- Ciphering Indicator for terminals with a suitable display.
  The ciphering indicator feature allows the ME to detect that ciphering 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. If this feature is not disabled by the SIM, then whenever a connection is in place, which is, or becomes unenciphered, an indication shall be given to the user. Ciphering itself is unaffected by this feature, and the user can choose how to proceed.

- Support for PLMN selection

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

14~~6~~ Charging principles~~ for UMTS~~

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 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.

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.).

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.

15~~7~~ 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 ~~UMTS~~ networks subject to appropriate roaming/commercial agreements.

~~Handover between UMTS and GSM systems (in both directions) is required, even if this requires changes to GSM specifications. In addition, a generic solution may be implemented in UMTS which allows calls to be handed over between UMTS and other pre-UMTS systems in both directions.~~

For further information see TS 22.129 [9].

16~~8~~ Network Selection

Three roles may be involved in ~~UMTS~~ network selection: the home environment, the serving network and the user. Services may be available to the user through a choice of several serving networks in a given location, possibly using different types of Radio Access Network, , however it is expected that a user terminal will communicate with one network at a given instant (there may be exceptions such as when an inter-network handover occurs).

All selection schemes make use of information provided by the serving networks, including the network name, the network capabilities and any restrictions. Other information such as terminal capabilities may also be required. This information may change with time but must be accurate and available at the time network selection is being made.

Procedures 1 and 2 below for network selection ~~in UMTS~~ shall be supported by all ~~mobile station~~user sequipment. The user shall be able to choose which procedure to use at any given time.

1. Default Automatic Procedure

A default procedure for network selection shall be defined which selects a network from amongst those available based upon information such as network name/ID, network capabilities, signal strength and network type. ~~The full list of parameters is FFS. The choice of which network to use should be developed from a procedure similar to that specified in GSM 02.11~~

2. Manual Procedure

The manual procedure consists in presenting to the user the list of all available networks and letting her make the selection. The user shall be able to make use of the manual selection procedure at any time.

3. Home Environment Specific Procedure

Optionally, if provisioned by the ~~MS~~UE and selected by the user, the home environment can add the ability to define the behaviour when selecting the required network from those available. A standardised framework for over-the-air transfer of behaviour definition is required. If enabled by the user, it shall be possible for the home environment procedure to instruct the ~~mobile station~~user equipment to search for a network which meets a given set of requirements, indicated by certain parameters or to compile a list of all available networks.

Other procedures may be offered by the ~~MS~~UE.

Both automatic and manual network selection schemes are constrained by commercial agreements between the home environments and serving networks. If a roaming agreement does not exist and cannot be established using the procedure for automatic establishment of roaming agreements, then registration on the network will not succeed. Therefore the user must provide sufficient information to allow the serving network to identify the relevant home environment and to allow the home environment to identify the user.

A ~~U~~SIM/USIM shall be registered on one and only one serving network at any given time (there may be exceptions such as when preparing for an inter-network handover). Changing the serving network between two calls requires SIM/USIM de-registration from the current serving network and SIM/USIM re-registration on the newly selected serving network. ~~If simultaneous access to more than one home environment is required (through a card with multiple USIMs or through several cards in a multi-slot terminal), manual selection shall be invoked.~~

A procedure for handling network rejections is required, see TS 22.001~~GSM 02.07 Annex A~~[4] for an example. During manual selection the user may be allowed to attempt to select any available network (subject to restrictions that may be specified by the home environment). Successful manual network selection shall update the procedure for preventing unnecessary network access attempts. A set of network access rejection causes shall be standardised.

It shall be possible for a network operator to control access to the network. This may be done for example to provide priority to emergency services or to control demand upon the access channel after a network failure, see ~~GSM~~TS ~~0~~22.011 [11] for further details.

17 Security

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

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.

A.4 Keypad

A physical means of entering numbers, generally, though not necessarily, in accordance with the layout shown in figure A.1.
See also 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.5 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 TS 22.003 [14].

A.6 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.7 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.8 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.

A.9 Dual Tone Multi Frequency (DTMF)

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

A.10 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.11 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.12 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 TS 22.003 [14]

A.13 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.14 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.15 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 TS 22.042 [15].

A.16 Network's indication of alerting in the UE~~MS~~

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.

8 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.17 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.18 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 SIM/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.19 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 shall not allow the making or receiving 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.20 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 GSM 04.08 [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 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.21 Selection of directory number in messages

The Short Message (Point to Point MT or MO, or Cell Broadcast), 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. This can be indicated by enclosing the directory number in a pair of inverted commas (" ").

If the displayed message contains these characters enclosing a directory number, a call can be set up by user action. Normal (unspecified) or International format (using + symbol) may be used.

The message may contain more than one directory number, in which case it is for the user to select the one required.

A.22 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.23 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.

Annex B~~A~~ (informative): Change history

3G CHANGE REQUEST

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

22.105 CR 021 Current Version: 3.6.0

3G specification number ↑ *↑ CR number as allocated by 3G support team*

For submision to TSG SA#6 for approval X
list TSG meeting no. here ↑ for information

(only one box should be marked with an X)

Form: 3G CR cover sheet, version 1.0 *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/3GCRF-xx.rtf*

Proposed change affects: USIM X ME X UTRAN X Core Network X
(at least one should be marked with an X)

Source:	SMG1/ SA WG1
Date:	
Subject:	Mainly editorial update for GSM/3GPP use.
3G Work item:	

Category:
(only one category shall be marked with an X)

F	Correction	
A	Corresponds to a correction in a 2G specification	
B	Addition of feature	
C	Functional modification of feature	X
D	Editorial modification	

Reason for change: The transfer of GSM specifications for 3GPP has created need to clarify requirements specifically related PS and CS domain. This CR updates 22.105 to be applicable to both the GSM and 3G systems. Further new ch is included identifying differences between GERAN and UTRAN

Clauses affected: All clauses

Other specs affected:

Other 3G core specifications		→ List of CRs:
Other 2G core specifications		→ List of CRs:
MS test specifications		→ List of CRs:
BSS test specifications		→ List of CRs:
O&M specifications		→ List of CRs:

Other comments:

help.doc

<--------- double-click here for help and instructions on how to create a CR.

1 Scope

~~Pre-UMTS~~Existing systems have largely standardised the complete sets of bearer services, teleservices and supplementary services which they provide. ~~One major difference between UMTS and pre-UMTS systems is that~~3GPP specifications specify - service capabilities rather than services~~ are standardised for UMTS~~, allowing service differentiation and system continuity.~~–~~ This Technical Specification (TS) describes how and what kind of services the ~~UMTS~~ user has access to.

2 References

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

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

2.1 Normative references

[1] TS 22.001: "Principles of circuit telecommunication services supported by a Public Land Mobile Network (PLMN)".

[2~~1~~] ~~GSM 0~~TS 02.002: "~~Digital cellular telecommunications system (Phase 2+);~~ Circuit Bearer services supported by a ~~GSM~~ Public Land Mobile Network (PLMN)".

[3~~2~~] ~~GSM ~~TS 2~~0~~2.003: "~~Digital cellular telecommunications system (Phase 2+); ~~Circuit Teleservices supported by a GSM Public Land Mobile Network (PLMN)".

[4~~3~~] ~~GSM 0~~TS 22.004: "~~Digital cellular telecommunications system (Phase 2+);~~ General on supplementary services".

~~[4] GSM 02.42: "Digital cellular telecommunications system (Phase 2+); Network Identity and Timezone (NITZ); Service description; Stage 1".~~

[5~~5~~] ~~GSM 0~~TS 22.038~~43~~: "~~Digital cellular telecommunications system (Phase 2+);~~ SIM toolkit ~~upport of Localised Service Area (SoLSA); Service description; ~~Stage 1".

[6~~6~~] ~~GSM 02~~TS 22.057: "~~Digital cellular telecommunications system (Phase 2+);~~ Mobile Station Application Execution Environment (MExE); Service description; Stage 1".

[7] TS 22.060: "General Packet Radio Service (GPRS) stage 1".

~~7] GSM 02.71: "Digital cellular telecommunications system (Phase 2+); Location Services (LCS); Service definition - Stage 1".~~

[8] ~~GSM 0~~TS 22.078: "~~Digital cellular telecommunications system (Phase 2+);~~ Customised Applications for Mobile network Enhanced Logic (CAMEL); Service definition - Stage 1".

~~[9] GSM 02.90: "Digital cellular telecommunications system; Unstructured Supplementary Service Data (USSD) - Stage 1".~~

[9~~10~~] ~~GSM ~~TS 22.101: "~~Universal Mobile Telecommunications System (UMTS); Service aspects;~~ Service principles".

[10~~11~~] ~~GSM ~~TS 22.121~~0~~: "~~Universal Mobile Telecommunications System (UMTS);~~ Virtual Home Environment (VHE), Stage 1".

~~[12] GSM 23.10: "Universal Mobile Telecommunications System (UMTS); UMTS Access Stratum; Services and Functions".~~

[11] TS 22.135: "Multicall, stage 1".

2.2 Informative references

[12~~1~~] ITU-T recommendation F.700: "Framework recommendation for audio-visual/multimedia services".

~~[2] GSM 02.01: "Digital cellular telecommunications system (Phase 2+); Principles of telecommunication services supported by a GSM Public Land Mobile Network (PLMN)".~~

3 Definitions and abbreviations

3.1 Definitions

For the purposes of this TS, the following definitions apply:

Basic telecommunication service : this term is used as a common reference to both bearer services and teleservices.

Bearer service : is a type of telecommunication service that provides the capability of transmission of signals between access points.

Call : a logical association between several users (this could be connection oriented or connection less).

Connection : is a communication channel between two or more end-points (e.g. terminal, server etc.).

Mobile termination : the mobile termination is the component of the ~~mobile station~~user equipment which supports functions specific to management of the radio interface (Um).

Multimedia service : Multimedia services are services that handle several types of media. For some services, synchronisation between the media is necessary (e.g. synchronised audio and video). A multimedia service may involve multiple parties, multiple connections, and the addition or deletion of resources and users within a single call.

Nomadic Operating Mode : Mode of operation where the terminal is transportable but being operated while stationary and may in addition require user co-operation (e.g. close to open spaces, antenna setup...).

Quality of Service : the collective effect of service performances which determine the degree of satisfaction of a user of a service. It is characterised by the combined aspects of performance factors applicable to all services, such as;

service operability performance;

- service accessibility performance;
- service ~~retainability~~ retention performance;
- service integrity performance; and
- other factors specific to each service.

Service Capabilities: Bearers defined by parameters, and/or mechanisms needed to realise services. These are

within networks and under network control.
Service Capability Feature: Functionality offered by service capabilities that are accessible via the standardised application interface
Services: Services are made up of different service capability features.
Supplementary service : is a service which modifies or supplements a basic telecommunication service. Consequently, it cannot be offered to a user as a standalone service. It shall be offered together with or in association with a basic telecommunication service. The same supplementary service may be common to a number of basic telecommunication services.

Teleservice; is a type of telecommunication service that provides the complete capability, including terminal equipment functions, for communication between users according to standardised protocols and transmission capabilities established by agreement between operators.

3.2 Abbreviations

For the purposes of this TS, the following abbreviations apply;

BER	Bit Error Rate
~~B-ISDN~~	~~Broadband ISDN~~
CAMEL	Customised Application for Mobile network Enhanced Logic
DTMF	Dual Tone Multiple Frequency
TR	Technical Report
TS	Technical Specification
ETSI	European Telecommunications Standards Institute
FAX	Facsimile
FER	Frame Erasure Rate
GSM	Global System for Mobile Communications
GERAN	GSM / EDGE Radio Access Network.
HE	Home Environment
~~IMUN~~	~~International Mobile User Number~~
IN	Intelligent Network
ISDN	Integrated Services Digital Network
~~ISO~~	~~International Organisation for Standardisation~~
ITU	International Telecommunication Union
LCS	Location Services
MExE	Mobile station Execution Environment
MMI	Man Machine Interface
MO	Mobile Origination
~~MS~~	~~Mobile Station~~
MT	Mobile Termination
~~O&M~~	~~Operations and Maintenance~~
~~PBX~~	~~Private Branch eXchange~~
PC	Personal Computer
~~PCMCIA~~	~~Personal Computer Memory Card International Association~~
PIN	Personal Identity Number
PNP	Private Numbering Plan
POTS	Plain Old Telephony Service
QoS	Quality of Service
USIM	User Service Identity Module
SMS	Short Message Service
SAT	SIM Application Toolkit
SN	Serving Network
SoLSA	Support of Localised Service Area
UE	User Equipment
~~UMTS~~	~~Universal Mobile Telecommunications System~~

4 Framework for the description of telecommunication services and applications

4.1 General

Telecommunication services ~~supported~~ defined by ~~UMTS~~ 3GPP specifications are the communication capabilities made available to users by home environment and serving network. A ~~UMTS network~~PLMN provides, in co-operation with other networks, a set of network capabilities which are defined by standardised protocols and functions and enable telecommunication services to be offered to users.
A service provision by a HE/SN to a ~~UMTS~~ user may cover the whole or only part of the means required to fully support the service.
The service classification and description which follow are independent of different possible arrangements for the ownership and provision to the user of the means required to support a service.

4.2 Basic telecommunication services

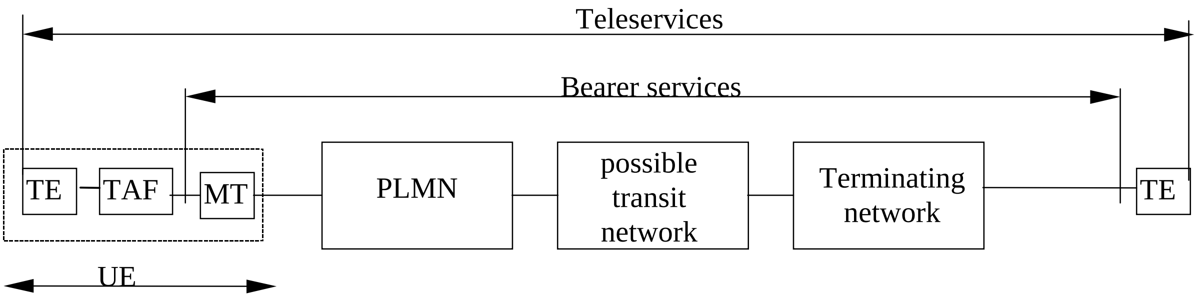
Basic telecommunication services are divided in two broad categories;

- bearer services, which are telecommunication services providing the capability of transmission of signals between access points;

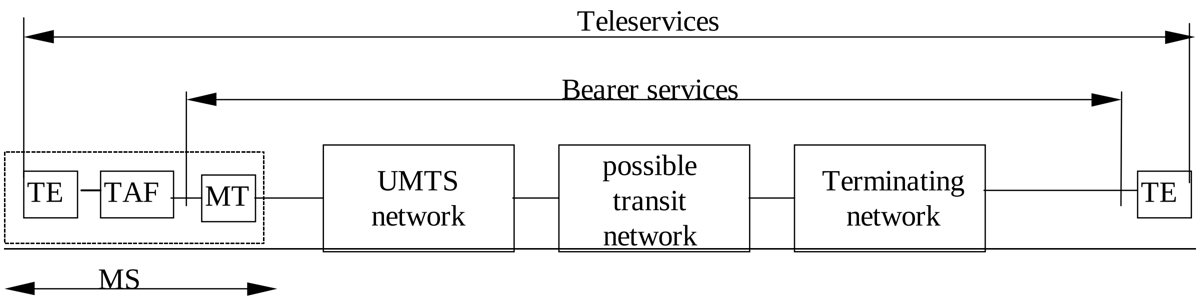
- teleservices, which are telecommunication services providing the complete capability, including terminal equipment functions, for communication between users according to protocols established by agreement between network operators.

The communication link between the access points may consists~~s~~ of– ~~UMTS~~PLMN, one or more transit networks and a terminating network. The networks between the two access points typically use different means for bearer control.

Figure 1 illustrates these definitions.



UE: User Equipment
MT: Mobile Termination
TE: Terminal Equipment
TAF: Teminal Adaption Function



MS: Mobile Station
MT: Mobile Termination
TE: Terminal Equipment
TAF: Teminal Adaption Function

NOTE 1: In order to limit the complexity of the figure, only one transit network is shown.

NOTE 2: The terminating network type may include a ~~UMTS network~~PLMN, either the originating one or another one.

NOTE 3: – The bearer service terminates in the ~~mobile station~~user equipment.

NOTE 4: – -The terminating network may be ~~a non UMTS~~another network such as: PSTN, ~~N-~~ISDN, ~~GSM,~~ IP networks/LANs and X.25

Figure 1; Basic telecommunication services supported by a ~~UMTS network~~PLMN

4.2.1 Bearer services

The characterisation of a bearer service is made by using a set of characteristics that distinguishes it from other bearer services. Particular values are assigned to each characteristic when a given bearer service is described and defined.
The service characteristics as they apply at a given reference point where the user accesses the bearer service.
In the general case networks between the two access points use different control mechanisms. In this case the bearer services of each network throughout the communication link have to be translated at the network interfaces to realize an end to end bearer service.
A list of definitions of attributes and values used for bearer services is contained in ~~clause~~chapter 5.
The bearer services are negotiable and can be used flexibly by applications.

4.2.2 Teleservices

Section 6 defines both standardised and non-standardised teleservices. Some teleservices are standardised because that interworking with other systems have been recognised as a requirement. Other teleservices will not be standardised. A decoupling between lower layer (i.e. bearer attributes) and higher layer capabilities will be necessary for the development of teleservices.

4.3 Supplementary services

A supplementary service modifies or supplements a basic telecommunication service. Consequently, it cannot be offered to a user as a stand alone service. It shall be offered together or in association with a basic telecommunication service. The same supplementary service may be applicable to a number of basic telecommunication services.

Two methods are used for the characterisation of supplementary services;

- The first method is used for the description of existing standardised supplementary services. These services are specified through the detailing of each of the operations involved in service provision and service usage (the provision/withdrawal, registration/erasure, activation/deactivation, invocation and interrogation operations). ~~Clause~~Chapter 7 lists these services.

- The second method enables the provision of HE/SN specific supplementary services. To make this possible, services can be built using service capability features which are accessed via the standardised application interface.

~~UMTS~~ A PLMN shall be able to handle multiple supplementary services within a call. Interactions shall be handled when several supplementary services are activated in the same call~~.~~. When multiple supplementary services can be activated concurrently, some prioritisation of the services will be necessary. Certain services may override or deactivate other services.

Interactions between operator specific supplementary services are not defined.

The following issues need consideration when interactions between services occur;

- Different phases of a call.

- A service spanning on more than one network.

- Service interactions that may occur between services offered to a single user, as well as between services offered to different interacting users.

NOTE: The methods defined for characterisation of services are description methods. They do not imply or restrict different implementations.

4.4 Service Capabilities

~~UMTS s~~Service capabilities– are based on functionality and mechanisms/toolkits such as provided by SAT [5], M~~e~~ExE [6], IN and CAMEL [8]~~.~~. These service capabilities– can be made visible to the– applications through an a~~the~~ pplication interface. See chapter 8 for service capability features.

5 Bearer Services

5.1 Definition of bearer services

Bearer services provide the capability for information transfer between access points and involve only low layer functions. These functions are sometimes referred as low layer capabilities (in reference to OSI layers). The user may choose any set of high layer protocols for his communication and the ~~UMTS network~~PLMN does not ascertain compatibility at these layers between users.

In the general case a communication link between access points provides a general service– for information transport. The communication link may span over different networks such as Internet, Intranets, LANs and ATM based transit networks, having network specific means for bearer control. Each network contributes to the end-to-end QoS perceived by the end-user.

PS and CS domains provide a specific set of bearer capabilities. The Circuit bearer services are described in 22.002 [2]. The packet services (GPRS) is described in TS 22.060 [7]. Following chapters describe the overall requirements for both the CS and PS domain bearers and also for the bearers used by teleservices.

5.2 Description of bearer services

Bearer services are characterised by a set of end-to-end characteristics with requirements on QoS. The characteristics and requirements shall cover major network scenarios, i.e. the cases when the terminating network is PSTN, ~~N-~~ISDN, ~~GSM,~~ IP networks/LANs, X.25 and a ~~UMTS network~~PLMN.

Quality of Service is the quality of a requested service (Teleservice or Bearer Service or any other service, e.g.

customer care) as perceived by the customer (ITU-T M.xxxx). QoS is always meant end-to-end. Network Performance of several network elements of the originating and terminating network(s) contribute to the QoS as perceived by the customer including terminals and terminal attachments. In order to offer the customer a certain QoS the serving network need to take into account network performance components of their network, reflect the performance of the terminal and ad sufficient margin for the terminating networks in case network performance requirements cannot be negotiated.

As far as the QoS to ~~3rd Generation~~ the subscriber is concerned ~~3G network~~network elements have to provide sufficient performance (reflecting possible performance constraints in terminating networks) so that the ~~3G networks~~PLMN cannot be considered as a bottleneck.

This section outlines the requirements on bearer services in two main groups;

- Requirements on information transfer, which characterise the networks transfer capabilities for transferring user data between two or more access points.

- Information quality characteristics, which describe the quality of the user information transferred between two or more access points.

It shall be possible to negotiate / re negotiate the characteristics of a bearer service at session / connection establishment and during an on going session / connection.

5.2.1 Information transfer

Connection oriented / ~~conectionless~~connectionless services

Both Connection oriented and connectionless services shall be supported.

Traffic type-

It is required that the bearer service provides one of the following:

- guaranteed/constant bit rate,
- non-guaranteed/dynamically variable bit rate, and
- real time dynamically variable bit rate with a minimum guaranteed bit rate..

Real time and non real time applications shall be supported.

- Real time video, audio and speech shall be supported. This implies the:
- ability to provide a real time stream of guaranteed bit rate, end to end delay and delay variation.
- ability to provide a real time conversational service of guaranteed bit rate, end to end delay and delay variation.
- Non real time interactive and file transfer service shall be supported. This implies the:
- ability to support message transport with differentiation as regards QoS between different users.
- Multimedia applications shall be supported. This implies the:
- ability to support several user flows to/from one user having different traffic types (e.g. real time, non real time)

Traffic characteristics

It shall be possible for an application to specify its traffic requirements to the network by requesting a bearer service with one of the following configurations

1) Point-to-Point
    - Uni-Directional
    - Bi-Directional
        - Symmetric
        - Asymmetric
2) Uni-Directional Point-to-Multipoint
    - Multicast

- Broadcast

A multicast topology is one in which sink parties are specified before the connection is established, or by subsequent operations to add or remove parties from the connection. The source of the connection shall always be aware of all parties to which the connection travels.
A broadcast topology is one in which the sink parties are not always known to the source. The connection to individual sink parties is not under the control of the source, but is by request of each sink party.

Note: Point-to-multipoint services are not supported by release 99 specifications.

In the case of a mobile termination with several active bearer services simultaneously, it shall be possible for each bearer service to have independent configurations and source/sink parties.~~5.2.2 Information quality~~

5.2.2 Information Quality

Information quality ~~a characterizes~~characterises the bit integrity and delay requirements of the applications.
Other parameters may be needed.

Maximum transfer delay

Transfer delay is the time between the request to transfer the information at one access point to its delivery at the other access point. In ~~clause~~chapter 5.5 requirements on maximum transfer delay is defined.

Delay variation
The delay variation of the information received information over the bearer has to be controlled to support real-time services. The possible values for delay variation are not a limited set, but a continuous range of values.

Bit error ratio
The ratio between incorrect and total transferred information bits. The possible values for Bit error ratio are not a limited set, but a continuous range of values.

Data rate
The data rate is the amount of data ~~tranferred~~transferred between the two access points in a given period of time.

5.3 Supported bit rates

It shall be possible for one application to specify its traffic requirements to the network by requesting a bearer service with any of the specified traffic type, traffic characteristics, maximum transfer delay, delay variation, bit error ratios & data rates. It shall be possible for the network to satisfy these requirements without wasting resources on the radio and network interfaces due to granularity limitations in bit rates.
It shall be possible for one mobile termination to have several active bearer services simultaneously, each of which could be connection oriented or connectionless.
The only limiting factor for satisfying application requirements shall be the cumulative bit rate per mobile termination at a given instant (i.e. when summing the bit rates of one mobile termination's simultaneous connection oriented and connectionless traffic, irrespective of the traffic being real time or non real time) in each radio environment:

- At least 144 kbits/s in satellite radio environment (Note 1).
- At least 144 kbits/s in rural outdoor radio environment.
- At least 384 kbits/s in urban/suburban outdoor radio environments.
- At least 2048 kbits/s in indoor/low range outdoor radio environment. (Note 2)

Note~~OTE~~ 1 : -This Peak Bit Rate may only be achieved in a nomadic operating mode.

Note 2: Not supported by GERAN.

5.4 Range of QoS requirements

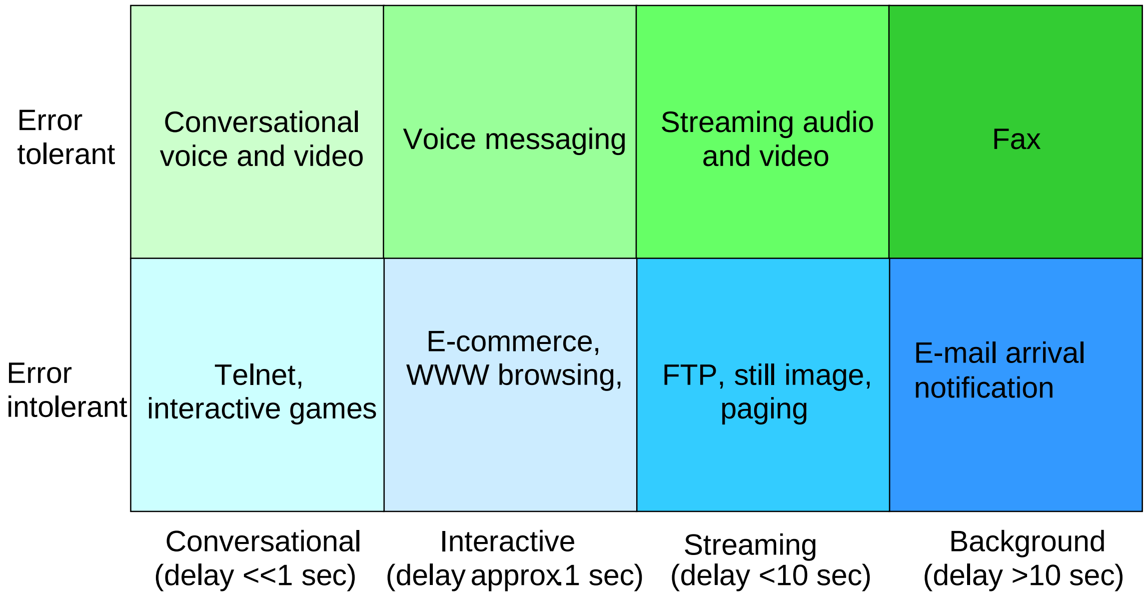
It shall be possible for one application to specify its QoS requirements to the network by requesting a bearer service with any of the specified traffic type, traffic characteristics maximum transfer delay, delay variation, bit error ratios & data rates.

The following table indicates the range of values that shall be supported~~ by UMTS~~. These requirements are valid for both connection and connectionless traffic. It shall be possible for the network to satisfy these requirements without wasting resources on the radio and network interfaces due to granularity limitations in QoS.

	Real Time (Constant Delay)	**Non Real Time (Variable Delay)**
Operating environment	**BER/Max Transfer Delay**	**BER/Max Transfer Delay**
Satellite (Terminal relative speed to ground up to 1000 km/h for plane)	Max Transfer Delay less than 400 ms BER 10-3 - 10-7 (Note 1)	Max Transfer Delay 1200 ms or more (Note 2) BER = 10-5 to 10-8
Rural outdoor (Terminal relative speed to ground up to 500 km/h) (Note 3)	Max Transfer Delay 20 - 300 ms BER 10-3 - 10-7 (Note 1)	Max Transfer Delay 150 ms or more (Note 2) BER = 10-5 to 10-8
Urban/ Suburban outdoor (Terminal relative speed to ground up to 120 km/h)	Max Transfer Delay 20 - 300 ms BER 10-3 - 10-7 (Note 1)	Max Transfer Delay 150 ms or more (Note 2) BER = 10-5 to 10-8
Indoor/ Low range outdoor (Terminal relative speed to ground up to 10 km/h)	Max Transfer Delay 20 - 300 ms BER 10-3 - 10-7 (Note 1)	Max Transfer Delay 150 ms or more (Note 2) BER = 10-5 to 10-8
NOTE 1; There is likely to be a compromise between BER and delay. NOTE 2; The Max Transfer Delay should be here regarded as the target value for 95% of the data. NOTE 3; The value of 500 km/h as the maximum speed to be supported in the rural outdoor environment was selected in order to provide service on high speed vehicles (e.g. trains). This is not meant to be the typical value for this environment (250 km/h is more typical).		

5.5 Supported End User QoS

This section outlines the QoS that shall be provided to the end user / applications. This section defines QoS requirements from end to end. The values in the tables are end to end, including mobile to mobile calls and satellite components. Figure 2 below summarises the major groups of application in terms of QoS requirements. Applications and new applications may be applicable to one more groups.



The following tables further elaborate ~~UMTS~~ end user / application QoS requirements.

Table 1: End-user Performance Expectations - Conversational / Real-time Services

Medium	**Application**	**Degree of symmetry**	**Data rate**	**Key performance parameters and target values**		
				One-way Delay	Delay Variation	Information loss
Audio	Conversational voice	Two-way	4-25 kb/s	<150 msec preferred <400 msec limit	< 1 msec	< 3% FER
Video	Videophone	Two-way	32-384 kb/s	< 150 msec preferred <400 msec limit Lip-synch : < 100 msec		< 1% FER
Data	Telemetry - two-way control	Two-way	<28.8 kb/s	< 250 msec	N.A	Zero
Data	Interactive games	Two-way	< 1 KB	< 250 msec	N.A	Zero
Data	Telnet	Two-way (asymmetric)	< 1 KB	< 250 msec	N.A	Zero

Table 2: End-user Performance Expectations - Interactive Services

Medium	**Application**	**Degree of symmetry**	**Data rate**	**Key performance parameters and target values**		
				One-way Delay	Delay Variation	Information loss
Audio	Voice messaging	Primarily one-way	4-13 kb/s	< 1 sec for playback < 2 sec for record	< 1 msec	< 3% FER
Data	Web-browsing - HTML	Primarily one-way		< 4 sec /page	N.A	Zero
Data	Transaction services – high priority e.g. e-commerce, ATM	Two-way		< 4 sec	N.A	Zero
Data	E-mail (server access)	Primarily One-way		< 4 sec	N.A	Zero

Table 3: End-user Performance Expectations - Streaming Services

Medium	**Application**	**Degree of symmetry**	**Data rate**	**Key performance parameters and target values**		
				One-way Delay	Delay Variation	Information loss
Audio	High quality streaming audio	Primarily one-way	32-128 kb/s	< 10 sec	< 1 msec	< 1% FER
Video	One-way	One-way	32-384 kb/s	< 10 sec		< 1% FER
Data	Bulk data transfer/retrieval	Primarily one-way		< 10 sec	N.A	Zero
Data	Still image	One-way		< 10 sec	N.A	Zero
Data	Telemetry - monitoring	One-way	<28.8 kb/s	< 10 sec	N.A	Zero

5.6 Radio Interface optimisation

The following requirements shall lead the radio interface optimisation process;

- support of high bit rate (around the Peak Bit Rate), bursty, asymmetric, non-real time bearer capabilities;

- support of high bit rate (around the Peak Bit Rate), bursty, asymmetric, real time bearer capabilities;

- the ability to extend or reduce the bandwidth associated ~~to~~ with a bearer capability in order to adapt to bit rate or radio condition variations and~~,~~ to add or drop service components.

However, the services provided by ~~GSM~~ existing systems (speech in particular) shall be supported in a spectrally efficient manner (at least as efficiently as in GSM) for the same quality of service.
In order to allow the support of flexible, bandwidth on demand services, bearer services should be provided with the finest possible granularity that can be efficiently supported.

~~5.7 Support of GSM general bearer services~~

~~UMTS shall support GSM General Bearer Services (GBS) and interworking scenarios as specified in 02.02.~~

6 Teleservices

6.1 Definition of teleservices

Teleservices provide the full capabilities for communications by means of terminal equipment, network functions and possibly functions provided by dedicated centres.

6.2 Description of teleservices

The basic reference ~~in~~ ~~UMTS~~ for the description of teleservices is the ITU-T F.700 [12] recommendation. F.700 provides a generic, network independent, description of multimedia services. The methodology used covers both single ~~mono~~media and multimedia services, the ~~mono~~ single media services being a particular type of multimedia services.
Multimedia services are classified into categories with similar functional characteristics. The six categories are multimedia conference services, multimedia conversational services, multimedia distribution services~~-~~, multimedia retrieval services, multimedia messaging services and multimedia collection services.
The rest of ~~clause~~chapter 6 describes the teleservices and options that shall be provided ~~by UMTS networks~~.
A teleservice can be viewed as set of upper layer capabilities utilising the lower layer capabilities described by the set of attributes in ~~clause~~chapter 5.
Multimedia teleservices support the transfer (and in some case retrieval, messaging, distribution) of several types of information (service components). For this reason, there are service attributes (relating to all the components of a teleservice) and service component attributes (relating to only one service component).

6.3 Support of teleservices ~~in UMTS networks~~

The realisation of teleservices requires the association of terminal and network capabilities. In the terminals and in the network, both upper layer capabilities and lower layer capabilities are necessary. The term upper layer capabilities is used because it relates to the OSI upper layers. Decoupling between upper layers and lower layers (transfer) is required. Even if this de-coupling may impact radio interface optimisation, it is nevertheless the only way of designing a system that is not outdated;

- Each time the information rate associated with an already supported teleservice is decreased by more efficient source coding techniques.

- Each time a new service is introduced that requires transfer capabilities not used by currently available teleservices.

Taking the example of two application that exchange information through a teleservice, the upper layer capabilities can be located in various places;

- In the two terminals if the two applications are connected to a ~~UMTS network~~PLMN.

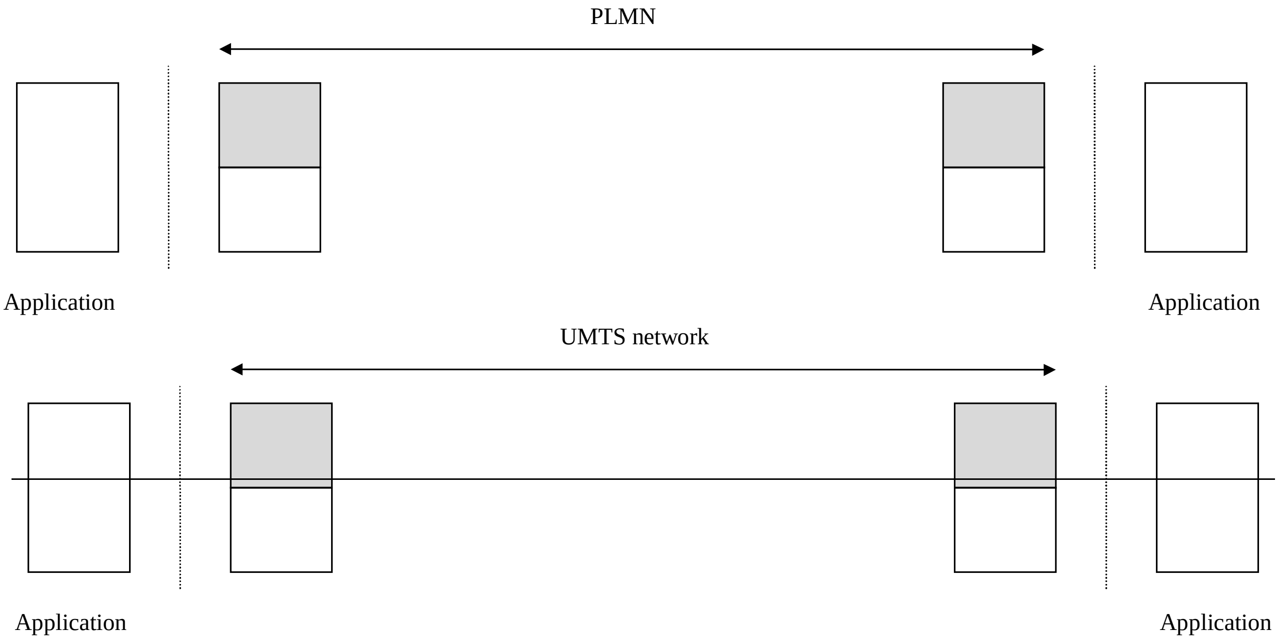


Figure 2; PLMN~~UMTS~~ teleservice

In the terminal of the application connected to a ~~UMTS network~~PLMN and in the upper layer interworking unit that is at the border of the ~~UMTS network~~PLMN and the target network if one application is connected to a ~~UMTS network~~PLMN and the other one is connected to another type of system. The upper layer interworking unit makes the adaptation between the ~~UMTS network~~PLMN and the target network at a service level.

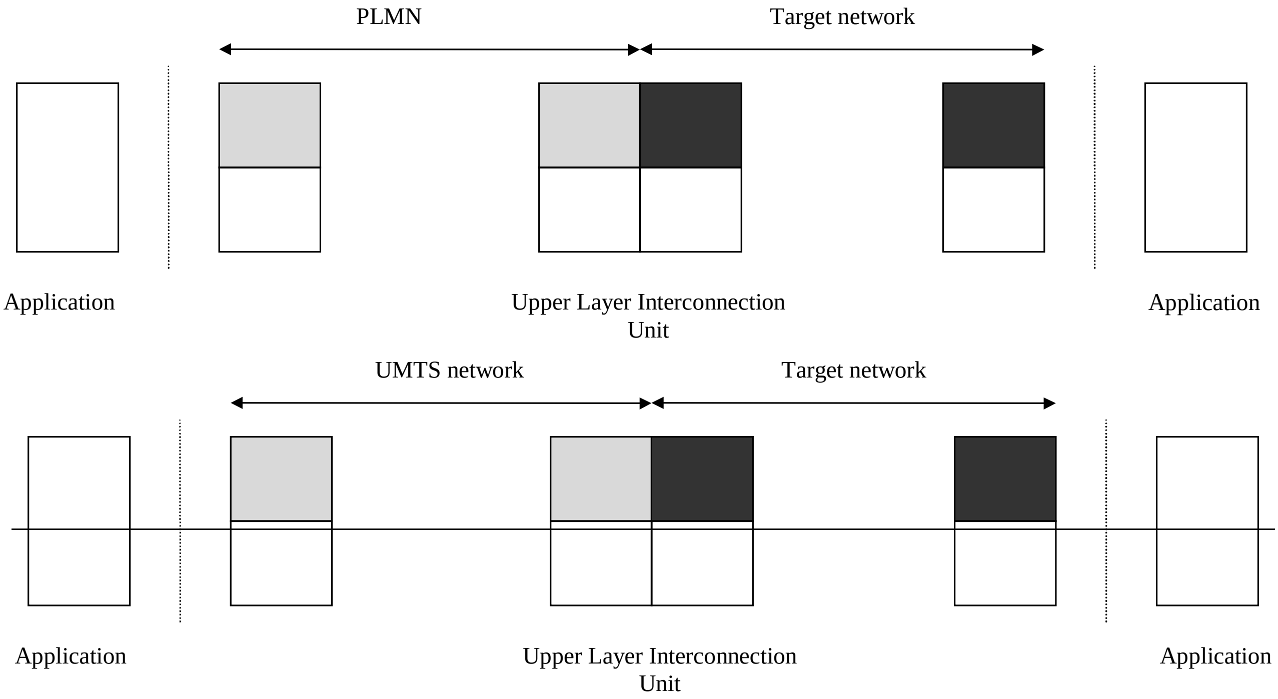


Figure 3; Teleservice with upper layer interworking

In the terminal of the application connected to a ~~UMTS network~~PLMN and in the terminal of the application connected to a target network if one application is connected to a ~~UMTS network~~PLMN and the other one is connected to another type of system, but only lower layer interconnecting unit is used at the border of the two networks. In this case, the interconnecting unit makes the adaptation between the ~~UMTS network~~PLMN and the target network at the transmission level.

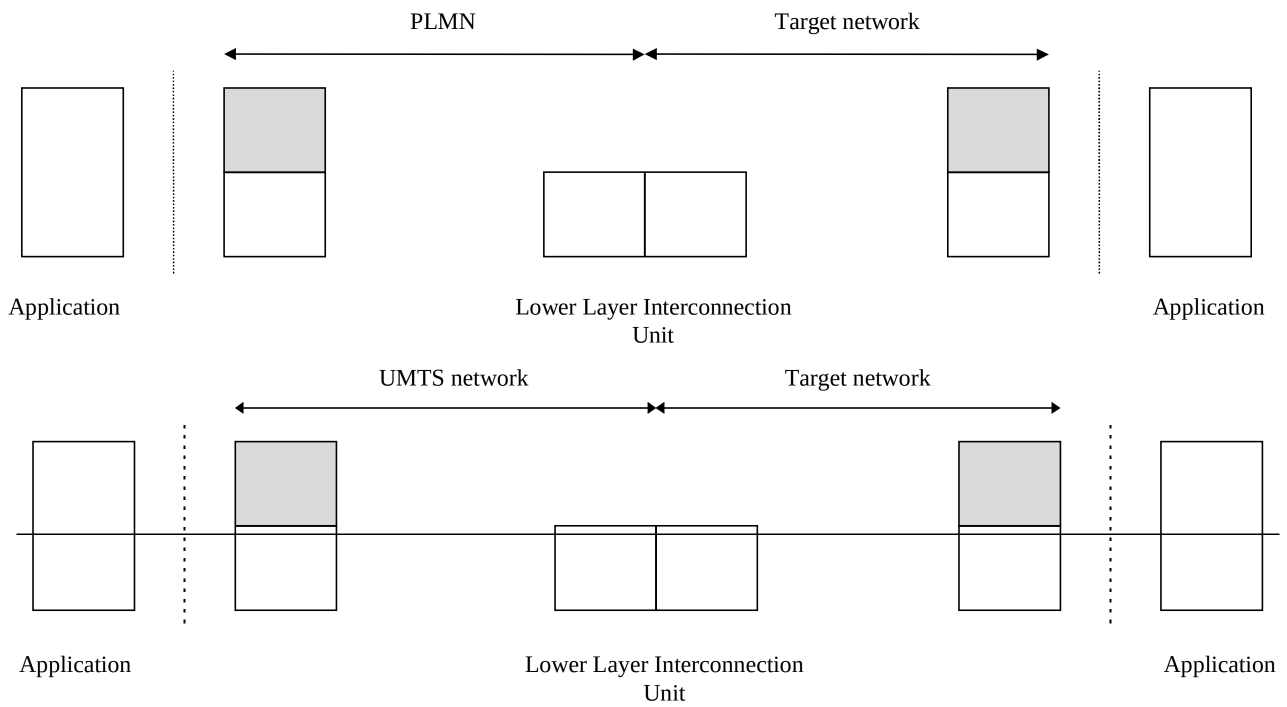


Figure 4; Teleservice with lower layer interworking

6.4 Existing Teleservices supported by ~~UMTS networks~~PLMN

The subset of standardised teleservices shall be supported ~~by UMTS~~ for interworking with teleservices provided on other networks. The means to support the following set of teleservices will be standardised;

- Speech;
- Emergency call;
- Short message service;

TS 22.003 [3] describes the circuit teleservices.

6.4.1 Speech

The speech service as defined in international standards should be supported~~ by UMTS~~. The international reference for the speech is ITU E.105 recommendation. ~~UMTS n~~Networks should contain interworking units which allow calls to be received from or destined to users of existing networks like PSTN or~~,~~ ISDN~~, GSM~~. This will include interworking units for generation of DTMF or other tones (the entire DTMF tone set would at minimum be available) and detection of DTMF tones.

A default speech codec shall be specified to provide speech service~~ across the UTRAN~~. The selected speech codec shall be capable of operating with minimum discernible loss of speech on handover between the GSM access network and UTRAN.

6.4.2 Emergency Call

This service will use a speech component. There are however compared to t~~T~~elephony reduced authentication requirements and a requirement for specific routing. Additionally Emergency Calls may have higher priority than normal calls~~, etc..~~… ~~The reference for the emergency call service is GSM 02.03.~~ See TS 22.101[9] for further details.

6.4.3 Short Message Service–- Point to Point (SMS-PP)

A~~The~~ short message service point to point ~~as specified in GSM 02.03~~ shall be supported~~ in UMTS~~. The~~A~~ short message service shall be provided seamlessly (as far as the user or the users terminal equipment is concerned) across the

U<ins>TRAN</ins>~~MTS~~ and GSM access network. ~~Additional features are planned for SMS in Release 99.~~

6.4.4 Short Message Service - Cell Broadcast (SMS-CB)

A short message service cell broadcast shall be provided seamlessly (as far as the user or the users terminal equipment is concerned) across the ~~UMTS~~ <ins>UTRAN</ins> and GSM network.

6.5 Internet Access

~~UMTS~~ <ins>3GPP specifications</ins> shall provide <ins>the</ins> means to interwork with external data networks. This interworking shall satisfy, within the constraints introduced by the mobile radio environment, the QoS requirements of the interworked-with network. ~~For UMTS t~~<ins>T</ins>he Internet is seen as the most important interworked-with network, therefore the specification of an optimised access to Internet will be part of the ~~UMTS standard~~<ins>3GPP specifications</ins>. The most important benefits achieved by the definition of Internet Access would be:

- Optimised transmission of IP traffic over the ~~UMTS~~ radio interface to minimise the amount of information transmitted.

- Optimised usage of encryption protocols/algorithms over the ~~UMTS~~ radio interface.

- Inter-operation of QoS mechanisms ~~used in both, UMTS and in Internet.~~

For the purposes of optimised access to Internet one or more of the ~~UMTS~~ generic bearers will be used. ~~On top of the bearer a UMTS protocol profile will be defined. This profile would be based on the work done by IETF or other relevant fora, and will consist of a recommended set of parameters and standardised protocols providing similar services than the Internet ones but optimised for wireless access. In the case of Internet traffic it would be possible for the user to select the encryption to be used (e.g. no encryption, end-to-end encryption, encryption over UMTS radio, etc.).~~ The QoS mechanisms defined for ~~UMTS~~ packet access mode will be harmonised with those defined for Internet (e.g. Differentiated Services).

7 Supplementary Services

Supplementary services are used to complement and personalise the usage of basic telecommunication services (bearer services and teleservices). The capabilities standardised ~~in UMTS~~ shall enable all the supplementary services specified in ~~GSM 02.04 and the 02.8x set~~<ins>TS 22.004 [4]</ins> to be provided.

8 Service Capability features

Services Capability Features are open, technology independent building blocks accessible via a standardised application interface. This interface shall be applicable for a number of different business and applications domains (including besides the telecommunication network operators also service provider, third party service providers acting as HE-VASPs, etc.).

All of these businesses have different requirements, ranging from simple telephony and call routing, virtual private networks, fully interactive multimedia to using ~~MS~~<ins>UE</ins> based applications.

The service capability features shall enable applications to make use of the service capabilities (e.g. CAMEL, MExE, etc) of the underlying ~~UMTS~~ network in an open and secure way.

Application/Clients access the service capability features via the standardised application interface. This means that a single service capability feature is accessible and visible to application/clients via the method/operation invocations in the interface.

Two different types of service capability features can be distinguished:

- **Framework service capability features**: these shall provide commonly used utilities, necessary for the non-framework service capability features to be accessible, secure, resilient and manageable.
- **Non-Framework service capability features**: these shall enable the applications to make use of the functionality of the underlying network capabilities (e.g. User Location service capability features).

<ins>For further information see TS 22.121 [10].</ins>

~~**Reference should be made to TS 22.121**~~

8.1 Framework service capability features

Framework service capability features will be used e.g. for authentication, registration, notification, etc. and provide functionality that is independent of any particular type of service. Other commonly used service capability features may be added later.

Examples of Framework Service Capability features are (TS 22.~~121~~ 121[10] ~~session 10.1~~):

- Authentication
- User-Network Authentication
- Application-Network Authentication
- User-Application Authentication
- Authorisation
- Application-Network Authorisation
- User-Application Authorisation
- Registration
- Discovery
- Notification.

8.2 Non-Framework service capability features

The Non-Framework service capability features represent the total collection of service capability features that are not included in the Framework. These non-framework service capability features~~–~~ enable the application to make use of the functionality provided by the network and service capabilities.
Service capability features shall be defined as much as possible in a generic way to hide the network specific implementation. To achieve this, it is necessary to identify the functionality that is provided by more than one service capabilities. For example, User Location can be produced in several underlying ways. This functionality can be captured once when defined the service capability features in a generic way. It is important that the generic part becomes as large as possible.
When applications use the generic service capability features, these applications become independent of (portable over) underlying service capabilities. Applications shall however still be able to request service capability features specific to a service capability (e.g. Call Setup from CAMEL). This will increase dependency of the used service capability.

Examples of Non-Framework service capability features are (TS 22.121[10] ~~session 10.2~~):

- Session Control
- Security/Privacy
- Address Translation
- Location

  The precision of the location shall be network design dependent, i.e. an operator choice. This precision may vary from one part of a network to another. It may be chosen to be as low as hundreds of meters in some place and as accurate as 5 meters in other place. It is required that a minimum precision of around 50 meters can be achieved in all types of terrestrial radio environment. Technical issues may constrain the precision to be mobile state dependent as well (mobile idle / mobile in communication). Several design optional features (e.g. size of the cell, adaptive antenna technique, path loss estimation technique...) shall allow the network operator to reach cost effectively the target precision.

  Because there may be very different uses of the location information;

  - It shall be possible to make the information available to the user, HE/SN and value added service providers. The user shall be able to restrict access to the location information (permanently or on a per call basis). The restriction can be overridden by the network operator when appropriate (e.g. emergency calls).

  - It shall be possible to set the delay to get the location information (the situation is quite different whether the information is needed for call routing or if it is needed by a user application).

  - It shall be possible to select the frequency of the location information update.

- to identify and report when the user’s terminal enters or leaves a specified geographic area.

- It shall be possible to specify the area as a circular zone (centre and radius) to a resolution that will be limited by the accuracy capability of the part of the serving network where the user is registered.

- User Status
- Terminal Capabilities
- Messaging
- Data Download

- User Profile Management
- Charging

9 Standardised Protocols and Capabilities

This ~~clause~~chapter introduces a list of standardised protocols and capabilities that shall be supported ~~by UMTS~~ for the control and creation of services. The access protocols and the execution environment described below are essential ~~for UMTS~~.

9.1 Access protocols

The access protocols shall allow the support of multimedia services. These services are characterised by the ability to dynamically change the number of participants and the number of connections during a call. The characteristics of the connections (confer the list of attributes used to describe a connection) may differ from one connection to another. They are negotiated during call set-up. They may be independently and dynamically re-negotiated on application (the telecommunication requirements of the application changes) or network initiative (change of network load conditions, during a handover procedure) during the call.

The application may require synchronisation between some of the connections. Later, this synchronisation shall not be lost during handover procedures.

Whenever a call is terminated in other types of networks, the negotiation shall take into account the limitations of these networks. Interworking shall be possible with PLMN, PSTN, ~~GSM,~~ ISDN and Internet networks. ~~Later releases will specify interworking with B-ISDN networks.~~

The access protocols shall allow a ~~mobile station~~user equipment to have several calls active simultaneously .

9.2 Execution Environment

The execution environment is a set of standardised capabilities that shall allow the support of HE/SN specific services (i.e. both applications, teleservices and supplementary services). The execution environment shall be distributed between the IC card, terminal and network nodes. The terminal and the serving network capabilities shall be the only limiting factor for the support of the services designed to run on the execution environment. The execution environment is composed of the following building blocks;

- A standardised content description language for support of HE/SN~~NO/SP~~ specific user interfaces (both for information output and user input). This is intended only for platforms which are terminals.

- A standardised procedural language for support of ~~NO/SP~~HE/SN specific scripts. This language shall be common to all types of platforms. The scripts could be used for e.g. improving the user interface, adding new features to the terminal like the latest version of a codec, controlling the execution of a service.

- Standardised application programming interfaces for opening platform resources and capabilities to the scripts written with the standardised procedural language. These interfaces would be platform type dependent. The interfaces shall include primitives for accessing to the basic control functions, as illustrated on the figures 5 and 6 below.

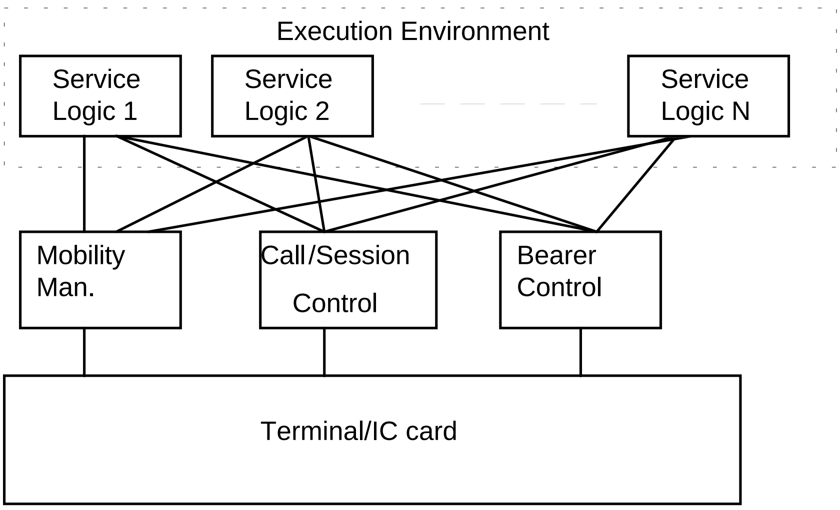


Figure 5: Execution Environment in the ~~Mobile Station~~User equipment

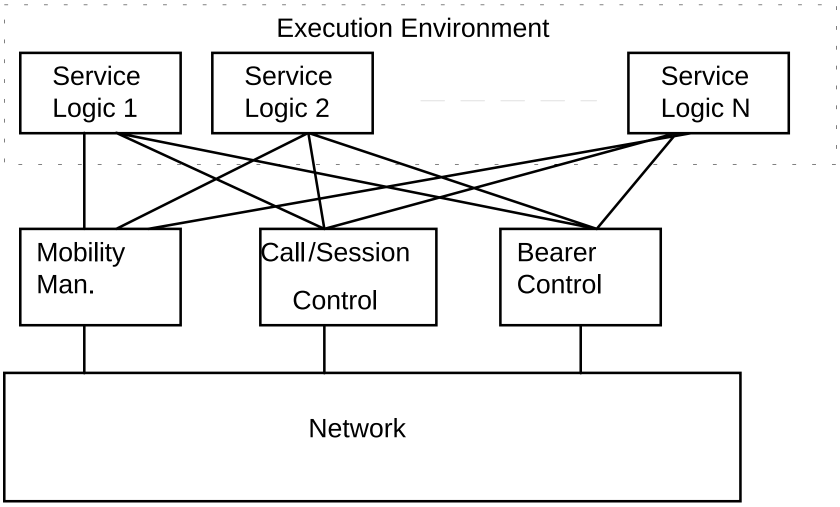


Figure 6: Execution Environment in the Network

- Call states, messages, information elements, values of information elements shall serve as triggers for subsequent interaction with service logic. The list of triggers shall incorporate those provided by ~~is for further study and is likely to incorporate~~ CAMEL, SIM Toolkit, MExE.
- Means to turn triggers on and off, and associate them with service logic will be standardised.
- A standardised certification scheme and security model with several levels of trusts in order to control the scripts access rights to the platform resources and capabilities. This would be used to allow e.g. the ~~SP~~ HE and the HE~~SP~~ only to access to SIM/USIM data.
- Standardised protocols for allowing the download of content description pages and scripts in the platform.

~~10 Existing GSM System features~~

~~Following GSM system features shall be supported by G-UMTS standards (for networks based on GSM evolution).~~

~~10.1 Network Identity and Time Zone (NITZ)~~

~~NITZ is specified in GSM 02.42.~~

~~10.2 Support of Localised Service Area (SoLSA)~~

~~SoLSA is specified in GSM 02.43.~~
~~Note : SoLSA modifications due to UTRAN related aspects are FFS.~~

~~10.3 Mobile station Execution Environment (MExE)~~

~~MExE is specified in GSM 02.57.~~

~~10.4 Location Services (LCS)~~

~~LCS is specified in GSM 02.71.~~
~~Note : LCS modifications due to UTRAN related aspects are FFS.~~

~~10.5 Customised Application for Mobile network Enhanced Logic (CAMEL)~~

~~CAMEL is specified in GSM 02.78.~~

~~10.6 Unstructured Supplementary Service Data (USSD)~~

~~USSD is specified in GSM 02.90~~
~~Note : USSD modifications due to UTRAN related aspects are FFS.~~

10 Access dependent services and features

This section describes the features that will be ~~dependant~~dependent on the mode of radio access.
In general different access networks provide different capabilities with different QoS.

- Multicall, as specified in TS 22.135 [11], is supported only via UTRAN.
- Packet switched traffic using GPRS over GERAN will have a maximum rate in the order of 384kb/sec.
- Packet switched traffic using UTRAN will have a maximum rate in the order of 2Mb/sec.
- ASCI teleservices, TS 22.003 [3] are only available in GERAN.
- The accuracy of the determination of location may differ between the various access technologies.
- At GERAN reception of CBS messages for a UE is not supported if it is connected in the CS domain or in the PS domain when data is currently transmitted.
- Transparent (T) fax as specified in TS 22.003 [3] is only supported by UTRAN.
- Non-transparent fax (NT) as specified in TS 22.003 [3] is only supported by GERAN.

Note: Differences of SoLSA features between GERAN and UTRAN is FFS.

Annex A (informative): Examples of services built from service capabilities features

Call Barring
In standard GSM, the Call Barring services allow to prevent outgoing calls to certain sets of destinations, based on the number dialled and whether the user is roaming. I~~n UMTS, i~~t is proposed that this service allows to block outgoing calls based on a wider range of parameters which could include factors such as the time of day, day of week, location, type of call requested, cost of the service and/or destination. This would allow to develop Call Barring services tailored to business and personal markets to avoid abuse.
This service is invoked during the initial outgoing call set-up procedure and allow the call to be blocked prior to incurring any charges. This Service can be applied to any teleservice for both connection-oriented and connectionless-oriented services.

Call Filtering/Forwarding
In standard GSM, there is no call filtering service. All calls are presented to the user unless a call forwarding service is used to re-direct calls; there is no different call handling depending on the incoming call parameters (although differentiation on call type (voice/data) is possible).
~~In UMTS, the~~The call filtering service allows the control of whether incoming calls are accepted, forwarded or terminated. The parameters which can be used to determine the final destination of a call may include the caller ID (CLI), original number dialled, time of day, current user location/network, user profile settings and current state of the terminal.
This service shall be two-stage; immediate call filtering (handled regardless of whether the terminal is online or not) and late call filtering (handled only if the terminal is online). It shall be possible to create and operate new call filtering services which can access any of the key parameters to handle calls in this way.

Hold
This service allows an established call to be maintained, whilst suspending use of the bearer from the incoming access point of the network. This saves on both air interface and network traffic resources when a call is temporarily suspended. The incoming access point in the network means either the originating ~~UMTS~~ terminal, or interworking point with another network.

Transfer
This service allows either an established or held call to be redirected to another destination. This may either be used by setting up a new call to the destination first, or simply redirecting the existing call to the new destination. It shall be possible to revert such a call back to the diverting terminal at any time before it is accepted (answered) by the new destination. The ~~UMTS~~ system shall ensure that an optimal traffic route is used after the call has been answered by its new (final) destination.

Call-back When Free
This service can be invoked where a call (or a connectionless message) cannot be delivered to its destination because it is in use. The~~ UMTS~~ system shall inform the requesting entity when the destination is next able to accept the call, allowing a new call to be originated. This allows existing GSM services, such as ~~Call-back When Free~~CCBS to be implemented. Where multiple requests are outstanding for a terminal which becomes available, the system shall determine in which order the requests are handled, probably in a serial manner. Ideally, it shall be possible to create the service logic which determines the order used from a range of accessible parameters.

Annex B (informative) : Description and analysis of communication schemes

This annex gives a high level classification and description of communications requirements from end users and applications.

B.1 Communication schemes

The requirements on bearer services are based on an analysis of user and application needs. Four end-user groups are identified according to four distinctly different communication schemes; Conversational - real time, Interactive services, Streaming services and Background services.

B.2 QoS related performance requirements for example end user applications

A typical user is not concerned with how a particular service is provided. However, the user is interested in comparing one service with another in terms of universal, user-oriented performance parameters which apply to any end-to-end service. From a user's perspective, performance should be expressed by parameters which:

- Focus on user-perceivable effects, rather than their causes within the network
- Are independent of the networks internal design
- Take into account all aspects of the service from the user's point of view which can be objectively measured at the service access point
- Can be assured to a user by the service providers(s)

With these considerations in mind, this section examines the requirements of typical end user applications that can be expected~~ in UMTS~~.

B.2.1 Performance requirements for conversational real-time

The most well known use of this scheme is telephony speech (e.g. GSM), but with Internet and multimedia a number of new applications will require this scheme, for example voice over IP and video conferencing tools. Real time conversation is always performed between peers (or groups) of live (human) end-users. This is the only scheme where the required characteristics are strictly given by human perception (the senses). Therefore this scheme raises the strongest and most stringent QoS requirements.

The real time conversation scheme is characterised by that the transfer time shall be low because of the conversational nature of the scheme and at the same time that the time relation (variation) between information entities of the stream shall be preserved in the same way as for real time streams. The maximum transfer delay is given by the human perception of video and audio conversation. Therefore the limit for acceptable transfer delay is very strict, as failure to provide low enough transfer delay will result in unacceptable lack of quality. The transfer delay requirement is therefore both significantly lower and more stringent than the round trip delay of the interactive traffic case.

Real time conversation - fundamental characteristics for QoS:

- preserve time relation (variation) between information entities of the stream
- conversational pattern (stringent and low delay)

The resulting overall requirement for this communication scheme is to support conversational real time services with low transfer delay as given by the human perception. (There are less hard requirements on packet loss ratio.)

A real-time streaming application is one that delivers time-based information in real-time, where time-based information is user data that has an intrinsic time component. Video, audio and animation are examples of time-based information, in that they consist of a continuous sequence of data blocks that shall be presented to the user in the right sequence at pre-determined instants.

Conversational voice

Audio transfer delay requirements depends on the level of interactivity of the end users. To preclude difficulties related to the dynamics of voice communications, ITU-T Recommendation G.114– recommends the following general limits for one-way transmission time (assuming echo control already taken care of):

0 to 150 ms	preferred range [<30ms, user does not notice any delay at all, <100ms, user does not notice delay if echo cancellation is provided and there are no distortions on the link]
150 to 400 ms	acceptable range (but with increasing degradation)
above 400 ms	unacceptable range

The human ear is highly intolerant of short-term delay variation (jitter) it is therefore paramount that this is reduced as lower level as is practical. A limit as low as 1 msec is suggested as a target.

Requirements for information loss are influenced by the fact that the human ear is tolerant to a certain amount of distortion of a speech signal. It is has been suggested in studies that acceptable performance is typically obtained with frame erasure rates (FER) up to 3 %.

A connection for a conversation normally requires the allocation of symmetrical communication resources, with the average hold time of a call being in the region of 2 minutes.

Videophone

Videophone implies a full-duplex system, carrying both video and audio and intended for use in a conversational environment. As such, in principle the same delay requirements as for conversational voice will apply, i.e. no echo and minimal effect on conversational dynamics, with the added requirement that the audio and video must be synchronised within certain limits to provide “lip-synch” (i.e. synchronisation of the speaker's lips with the words being heard by the end user). In fact, due to the long delays incurred in even the latest video codecs, it will be difficult to meet these requirements.

Once again, the human eye is tolerant to some loss of information, so that some degree of packet loss is acceptable depending on the specific video coder and amount of error protection used. It is expected that the latest video codecs will provide acceptable video quality with frame erasure rates up to about 1%.

Interactive games

Requirements for interactive games are obviously very dependent on the specific game, but it is clear that demanding applications will require very short delays, and a value of 250 msecs is proposed, consistent with demanding interactive applications.

Two-way control telemetry

Two-way control telemetry is included here as an example of a data service which does require a real-time streaming performance. Clearly, two-way control implies very tight limits on allowable delay and a value of 250 msec is proposed, but a key differentiator from the voice and video services in this category is the zero tolerance for information loss (obvious if you are controlling an important industrial process, for example).

Telnet

Telnet is included here with a requirement for a short delay in order to provide essentially instantaneous character echo-back.

B.2.2 Performance requirements for Interactive Services

When the end-user, that is either a machine or a human, is on line requesting data from remote equipment (e.g. a server), this scheme applies. Examples of human interaction with the remote equipment are: web browsing, data base retrieval, server access. Examples of machines interaction with remote equipment are: polling for measurement records and automatic data base enquiries (tele-machines).
Interactive traffic is the other classical data communication scheme that on an overall level is characterised by the request response pattern of the end-user. At the message destination there is an entity expecting the message (response) within a certain time. Round trip delay time is therefore one of the key attributes. Another characteristic is that– the content of the packets must be transparently transferred (with low bit error rate).
Interactive traffic– - fundamental characteristics for QoS:

– request response pattern
– preserve payload content

The resulting overall requirement for this communication scheme is to support interactive non-real time services with low round-trip delay.

Voice messaging and dictation

Requirements for information loss are essentially the same as for conversational voice, but a key difference here is that there is more tolerance for delay since there is no direct conversation involved. The main issue, therefore becomes one of how much delay can be tolerated between the user issuing a command to replay a voice message and the actual start of the audio. There is no precise data on this, but a delay of the order of a few seconds appears reasonable for this application.

Data

Although there may be some exceptions, as a general rule it is assumed that from a user point of view, a prime requirement for any data transfer application is to guarantee essentially zero loss of information. At the same time, delay variation is not applicable. The different applications therefore tend to distinguish themselves on the basis of the delay which can be tolerated by the end-user from the time the source content is requested until it is presented to the user.

Web-browsing

In this category we will refer to retrieving and viewing the HTML component of a Web page, other components eg images, audio/video clips are dealt with under their separate categories. From the user point of view, the main performance factor is how fast a page appears after it has been requested. A value of 2-4 seconds per page is proposed, however improvement on these figures to a target figure of 0.5 seconds wound be desirable.

High-priority transaction services (E-commerce)

The main performance requirement here is to provide a sense of immediacy to the user that the transaction is proceeding smoothly. A value of 2-4 seconds is suggested to be acceptable to most users.

E-mail (server access)

E-mail is generally thought to be a store and forward service which in principle can tolerate delays of several minutes or even hours. However, it is important to differentiate between communications between the user and the local email server and server to server transfer. When the user communicates with the local mail server, there is an expectation that the mail will be transferred quite rapidly, although not necessarily instantaneously. Consistent with the research findings on delay tolerance for Web-browsing, a requirement of 2-4 seconds is proposed.

B.2.3 Performance requirements for streaming services

When the user is looking at (listening to)– video (audio) the scheme streams applies. The real time data flow is always aiming at a live (human) destination. It is a one way transport.

This scheme is one of the newcomers in data communication, raising a number of new requirements in both telecommunication and data communication systems. First of all it is a mainly unidirectional stream with high continuous utilisation (i.e. having few idle/silent periods.) It is also characterised by that the time relations (variation) between information entities (i.e. samples, packets) within a flow must be preserved, although it does not have any requirements on low transfer delay.

The delay variation of the end-to-end flow must be limited, to preserve the time relation (variation) between information entities of the stream. But as the stream normally is time aligned at the receiving end (in the user equipment), the highest acceptable delay variation over the transmission media is given by the capability of the time alignment function of the application. Acceptable delay variation is thus much greater than the delay variation given by the limits of human perception.

Real time streams - fundamental characteristics for QoS:

- unidirectional continuous stream
- preserve time relation (variation) between information entities of the stream

The resulting overall requirement for this communication scheme is to support streaming real time services having unidirectional data flows with continuous utilisation. (There are less stringent requirements on delay and packet loss ratio, i.e. the ratio of lost or corrupted packets out of all packets sent.)

Audio streaming

Audio streaming is expected to provide better quality than conventional telephony, and requirements for information loss in terms of packet loss will be correspondingly tighter. However, as with voice messaging, there is no conversational element involved and delay requirements can be relaxed, even more so than for voice-messaging.

One-way video

The main distinguishing feature of one-way video is that there is no conversational element involved, meaning that the delay requirement will not be so stringent, and can follow that of streaming audio.

Bulk data

This category includes file transfers, and is clearly influenced by the size of the file. As long as there is an indication that the file transfer is proceeding, it is reasonable to assume some what longer tolerance to delay than for a single Web-page.

Still image

This category includes a variety of encoding formats, some of which may be tolerant to information loss since they will be viewed by a human eye. However, given that even single bit errors can cause large disturbances in other still image formats, it is argued that this category should in general have zero information loss. However, delay requirements for still image transfer are not stringent, given that the image tends to be built up as it is being received, which provides an indication that data transfer is proceeding.

Telemetry (monitoring)

Monitoring covers a wide range of applications, but in this category it is taken to apply to relatively low priority activities, eg status updating, rather than control.

B.2.4 Performance requirements for Background– applications

When the end-user, that typically is a computer, sends and receives data-files in the background, this scheme applies. Examples are background delivery of E-mails, SMS, download of databases and reception of measurement records. Background traffic is one of the classical data communication schemes that on an overall level is characterised by that the destination is not expecting the data within a certain time. The scheme is thus more or less delivery time insensitive. Another characteristic is that the content of the packets must be transparently transferred (with low bit error rate).

Background traffic– - fundamental characteristics for QoS:

- the destination is not expecting the data within a certain time
- preserve payload content

The resulting overall requirement for this communication scheme is to support non-real time services without any special requirement on delay.

A background– application is one that does not carry delay information. In principle, the only requirement for applications in this category is that information should be delivered to the user essentially error free. However, there is still a delay constraint, since data is effectively useless if it is received too late for any practical purpose.

Fax

Fax is included in this category since it is not normally intended to be an accompaniment to real-time communication. Nevertheless, there is an expectation in most business scenarios that a fax will be received within about 30 seconds. The information loss requirement is based on established wireline requirements for a Group 3 fax. As for the symmetry this should provide the required through put in the sending direction and the control signalling in backwards direction, hence an asymmetric connection is required.

Low priority transaction services

An example in this category is Short Message Service (SMS). 30 seconds is proposed as an acceptable delivery delay value.

Email (server to server)

This category is included for completeness, since as mentioned earlier, the prime interest in email is in the access time. There is a wide spread in user expectation, with a median value of several hours.

B.3 Adaptability and bearer service negotiation

Applications using the interactive or real time conversational communication schemes can also be described according to their possibilities for adapting to different environmental conditions as follows:

- Rigid applications; these applications can not adapt at all (e.g. GSM full rate speech.)
- Adaptive applications; these applications can adapt to the environment; they therefore require the network to support service negotiation. (e.g. multi-rate speech codecs)
- Elastic applications; these applications adapt totally to the environment and do therefore not require service negotiation (e.g. web browsing.)

The resulting overall requirement is to support service negotiation.