ETSI/GSM

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Title: GENERAL REQUIREMENTS ON INTERWORKING BETWEEN THE PLMN AND THE ISDN OR PSTN.

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1. Scope

The purpose of this recommendation is to identify the IWFs and requirements to support interworking between :

i) PLMN and PSTN
ii) PLMN and ISDN

It is not possible to treat ISDN and PSTN as one type of network, even when both ISDN and PSTN subscribers are served by the same exchange because of the limitations of the PSTN subscribers access ie analogue connection without D-channel signalling.

Within this recommendation, the requirements for voice and non-voice (data) calls are considered separately.

Data calls can be classified into two broad categories:

i) Circuit-mode
ii) Packet-mode

2. Introduction

General Network Interworking Scenarios are described in GSM 09.01. Most calls initiated by a PLMN user will terminate in either a PSTN or an ISDN. Since the numbering plan for the ISDN era (E.164) includes the numbering plan for the telephone network (E.163), it is not possible to distinguish by the number whether a given subscriber is a PSTN or ISDN subscriber. Further, in some countries both PSTN and ISDN subscribers will be connected to the same exchange, so the only difference for this type of combined network will be in the nature of the customer access.

When two dissimilar networks are required to interwork in order to support a communication between two subscribers, one on each network, a number of Interworking Functions (IWFs) are required to support the communication. Some of these are related to the differences in signalling and are dealt with in GSM 09.03. Examples of other aspects of interworking are;

i) the need or otherwise of echo control devices
ii) The need or otherwise of modem pools and network-based rate adaptation

For the purposes of determining the required IWFs, it is necessary, however, to consider separately each type of interworking (ie PLMN-ISDN and PLMN-PSTN) since, in the worst case, "PSTN" could refer to an essentially analogue network with electromechanical switching not controlled by software and without common-channel signalling.

Some facilities associated with alternate speech and data may not be available with version 1 of the MAP. See appendix for details.

3. References

Bearer Services	Refer to GSM 02.02
Teleservices	Refer to GSM 02.03
Connection Types	Refer to GSM 03.10
Signalling interworking	See Rec. GSM 09.03
Numbering	See Rec. GSM 03.03
Supplementary service interworking	See Rec. GSM 03.11
Rate Adaptation	See Rec. GSM 04.21 & 08.20

4. Definitions

Use is made of the following terms within this recommendation. These terms refer to information requirements necessary to support interworking functions, some of these terms will be identifiable with their use in other recommendations.

- Bearer capability information

  Specific information defining the lower layer characteristics required within the network.

- Lower layer capability information

  Information defining the lower layer characteristics of the terminal.

- Higher layer capability information

  Information defining the higher layer characteristics of a teleservice active on the terminal.

- Protocol identifier

  Information defining the specific protocols utilised for the support of data transfer by a terminal.

- Progress indicator

  Information supplied to indicate to the terminal that network interworking has taken place.

- Out of band parameter exchange

  Information exchanged via an associated or non-associated signalling link e.g. SS No 7.

Term	Recommendation where defined CCITT	GSM
Bearer Service	I.112, I.210, I.211	02.02
Exchange	I.112	
Packet Assembly/Disassembly	X.15	
Terminal Adaptor (Function)	I.411	(07-series)
Interworking Function	X.300	
Attributes	I.112, I.130, I.211	
Connection capabilities	I.340	
D channel	I.412	
B channel	I.412	

5. Abbreviations

DP	Dial Pulse
DTE	Data Terminal Equipment
DTMF	Dual Tone Multiple Frequency
IDN	Integrated Digital Network
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part (of Signalling System No.7)
LE	Local Exchange
MSC	Mobile Switching Centre
NT	Network Termination
PABX	Private Automatic Branch Exchange
PLMN	Public Land Mobile Network
PSTN	Public Switched Telephone Network
SS No.7	Signalling System No.7
TE	Terminal Equipment
TA	Terminal Adaptor
TUP	Telephone User Part (of Signalling System No.7)
RLP	Radio Link Protocol
L2R	Layer 2 Relay Function
Cct	Circuit

6. Network Characteristics

6.1 Key Characteristics of Networks Concerned

Characteristic	GSM PLMN	ISDN	PSTN
Subscriber Interface	Digital	Digital	Analogue
User-network signalling	GSM 04.08	I.441/I.451	loop-disconnect and DTMF
User-terminal equipment supported	MT0, MT1 or MT2 functions (see rec. 04.02)	Digital TE (ISDN NT, TE1 or TE2+TA) see I.411	Analogue TE (eg. dial pulse telephones, PABXs modemequipped DTEs)
Inter-exchange signalling	SS No.7 ISUP TUP+, MAP	SS No.7 ISDN Userpart (ISUP),TUP+	Channel associated (eg. R2, No.4, No.5) or common channel (eg. No.6, No.7, TUP)
Transmission facilities	Digital	Digital	Analogue/digital
Exchange types	Digital	Digital	Analogue/digital
Information transfer mode	Circuit/Packet	Circuit/Packet	Circuit
Information transfer capability	Speech, digital unrestricted alternate speech/ group 3 fax etc.	Speech, digi- tal unrestric- ted 3.1 kHz, audio, video etc.	3.1 kHz audio (voice/voice- band data)

TABLE 1/09.07
Key Characteristics of Networks Concerned

6.1.1 Characteristics of PLMNs

The GSM PLMN is fully defined in the GSM series of recommendations.

6.1.2 Characteristics of PSTNs

Because of the efforts at an early stage to standardize ISDNs in different countries, the differences between any two ISDNs will be small compared with the differences between PSTNs, which have evolved in different ways in different countries. In some cases the evolution has occurred over many decades, and therefore each PSTN is distinct, and for a recommendation on interworking, it is necessary to make certain assumptions about a generalized PSTN.
Whilst the key characteristics of PSTNs are given in Table 1 above, the specific IWFs needed to allow interworking between a PLMN and a PSTN will depend on the nature of the PSTN concerned.

TABLE 2/09.07 below gives a number of categories that can be used to classify PSTNs and a number of possibilities within each category.

Category	Possibilities within Category
Type of subscriber signalling	a) PSTN with loop disconnect subscriber signalling (10 pps)
	b) PSTN with DTMF subscriber signalling
Type of interexchange signalling	a) PSTN with channel-associated signalling
	b) PSTN with common-channel signalling
Type of interexchange transmission	a) Analogue
	b) Digital
Type of exchange switching	a) PSTN with electro-mechanical switching
	b) PSTN with electronic (non-digital) switching
	c) PSTN with electronic digital switching
Type of exchange control	a) Non-SPC
	b) SPC

TABLE 2/09.07
Characteristics of PSTNs

Note: Under each category, it is possible that a PSTN will have a combination of the possibilities rather than only one.

6.1.3 Characteristics of ISDN

These are defined in the CCITT I-series recommendations

7. Interworking classifications

7.1 Service interworking

Service interworking is required when the Teleservices at the calling and called terminals are different (e.g. Teletex interworking with facsimile). No service interworking has been identified as a requirement of the GSM system for PSTN/ISDN network based services.

7.2 Network interworking

Network interworking is required whenever a PLMN and a non-PLMN together are involved to provide an end to end connection and may be required in instances of PLMN to PLMN connections.

The concept of Bearer Services was developed for the ISDN and has been extended to the PLMN. A bearer service is defined (in CCITT I.112 and GSM 02.01) as:

<u>A type of telecommunication service that provides the capability for the transmission of signals between user-network interfaces.</u>

Bearer services are described by a number of attributes, where an attribute is defined as a specified characteristic of an object or element whose values distinguish that object or element from others (I.130).

For the purpose of this recommendation, a PSTN is assumed to provide a bearer service which equates to an ISDN 3.1 kHz audio bearer service.

<u>Refer to GSM 02.02 for complete list of bearer service attributes. Refer to GSM 04.08 for coding of Bearer Capabilities.</u>

<table>
<tr><th>Bearer service category in GSM PLMN</th><th>Access at Mobile Station</th><th>Bearer service in ISDN</th><th>Bearer service in PSTN</th></tr>
<tr><td rowspan="6">Circuit mode unstructured with unrestricted digital capability

Transparent and Non-transparent</td><td>Data Cct duplex asynchronous 300 bit/s</td><td rowspan="10">Cct mode structured 64 kbit/s unrestricted or Cct mode 3.1 kHz audio</td><td rowspan="3">Cct mode 3.1 kHz audio</td></tr>
<tr><td>Data Cct duplex asynchronous 1.2 kbit/s</td></tr>
<tr><td>Data Cct duplex asynchronous 1200/75 bit/s</td></tr>
<tr><td>Data Cct duplex asynchronous 2.4 kbit/s</td><td rowspan="3">Not applicable (no asynchronous modems specified currently in CCITT nationallly they may be available)</td></tr>
<tr><td>Data Cct duplex asynchronous 4.8 kbit/s</td></tr>
<tr><td>Data Cct duplex asynchronous 9.6 kbit/s</td></tr>
<tr><td rowspan="4">Circuit mode unstructured with unrestricted digital capability. Transparent</td><td>Data Cct duplex synchronous 1.2 kbit/s</td><td rowspan="4">Cct mode 3.1 kHz audio</td></tr>
<tr><td>Data Cct duplex synchronous 1.2 kbit/s</td></tr>
<tr><td>Data Cct duplex synchronous 4.8 kbit/s</td></tr>
<tr><td>Data Cct duplex synchronous 9.6 kbit/s</td></tr>
<tr><td rowspan="6">Pad access

Transparent and non-transparent</td><td>Data Cct duplex asynchronous pad access 300 bit/s</td><td rowspan="6">Cct mode structured 64 kbit/s unrestricted or Cct mode 3.1 kHz audio</td><td rowspan="4">Cct mode 3.1 kHz audio</td></tr>
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<tr><td>Data Cct duplex asynchronous pad access 9.6 kbit/s</td></tr>
</table>

TABLE 3/09.07 - Bearer Service Interworking

Bearer service category in GSM PLMN	Access at Mobile Station	Bearer service in ISDN	Bearer service in PSTN
Packet mode service Transparent and non-transparent	Data packet duplex synchronous 2.4 kbit/s	Cct mode 64 kbit/s unrestricted (X.32 access) or virtual circuit bearer service	Cct mode 3.1 kHz audio
	Data packet duplex synchronous 4.8 kbit/s		
	Data packet duplex synchronous 9.6 kbit/s		
3.1 kHz Ex PLMN Transparent and non-transparent	Cct mode unrestricted digital 300 to 9600 bit/s	Cct mode 3.1 kHz audio	
Circuit mode unstructured with alternate speech unrestricted digital capability. Transparent and non-transparent	Cct mode alternate speech with digital access capa-bilities 300 to 9600 bit/s as above	Cct mode structured 64 kbit/s unrestricted	
Circuit mode with speech followed by unrestricted digital capabil. Transparent and non-transparent	Cct mode speech followed by digital access capabilities 300 to 9600 bit/s as above	Cct mode 3.1 kHz audio	

TABLE 3/09.07 - Bearer Service Interworking (cont'd)

Teleservice in GSM PLMN	Access at Mobile Station	Bearer service in ISDN	Bearer service in PSTN
Telephony	Unstructured with speech capability	Speech or Cct mode 3.1 kHz audio	Cct mode 3.1 kHz audio
Emergency calls	Unstructured with speech capability		
Advanced MHS access 2.4 to 9.6 kbit/s	Data packet duplex synchronous 2.4 to 9.6 kbit/s	VCBS or 64 kbit/s unrestricted digital	Cct mode 3.1 kHz audio
Videotex access profile 1	Data Cct duplex asynchronous access 1200/75 bit/s	Cct mode structured unrestricted digital 64 kbit/s	
Videotex access profile 2	Data Cct duplex asynchronous access 1200/75 bit/s		
Videotex access profile 3	Data Cct duplex asynchronous access 1200/75 bit/s		
Teletex access CS	Data Cct duplex synchronous access 2.4 kbit/s	Cct mode structured unrestricted digital 64 kbit/s	
Teletex access PS	Data packet duplex synchronous access 2.4 kbit/s	VCBS or Cct mode unrestricted digital 64 kbit/s	
Facsimile group 3	Data Cct duplex synchronous access access alternate speech group 3 fax	Cct mode 3.1 kHz audio	

TABLE 4/09.07
Network interworking of GSM Teleservices

This table does not identify any relationship between Teleservices in the GSM PLMN with those in the ISDN/PSTN, it is merely to identify the interworking of the lower network layers of that teleservice with the network layers i.e. bearer service in the ISDN/PSTN.

7.3 Signalling interworking

See Recommendation 09.03.

7.4 Numbering

See Recommendation 03.03.

7.5 Supplementary service interworking

See Recommendation 03.11.

8. Compatibility checking and indication of compatibility requirements

Compatibility checking is carried out on the following items:-

a) Low layer compatibility - utilising lower layer and bearer capability information elements.

b) High layer compatibility - utilising higher layer capability information element.

Indication of compatibility requirements is carried out as described in section 9.2.2 under "Functional operation" and "b) Mobile subscriber indicates requirement in call confirmation message" and section 10.2.2 "Network interworking mobile terminated".

9. Interworking to PSTN

9.1 Speech Calls

9.1.1 Interworking indications to PLMN terminal. An indication to inform the PLMN terminal that:

i) instead of receiving out-of-band indications for certain types of failure conditions, a tone or announcement will be received from the PSTN.

ii) there will be a limitation on service selection information and address - the terminal may be required to accept the call without out-band compatibility checking.

iii) (if a DTE) in-band handshaking signals should be anticipated.

9.1.2 Network Indication of Change of Service

For a PLMN or ISDN user, the network will always provide an indication to the user of change of service at the user interface. The change of service may be due to the following reasons:

i) interworking with another network of reduced capability eg PSTN.

ii) resource constraints in the network.

In addition to providing an indication, the network will solicit user acceptance of the change of service in certain cases.

Examples are:

i) downgrading of service

ii) upgrading of service

9.1.3 Transmission aspects

Includes control of Speech Processing and Echo Control Devices, see GSM recommendation 03.50.

9.1.4 Generation of In-band Tones and Announcements (PLMN-PSTN)

In-band tones and announcements shall be provided for all speech and 3.1 kHz audio bearer services between a PLMN and a PSTN.

9.2 Data Calls

In this case a modem will be utilised to provide the interworking function

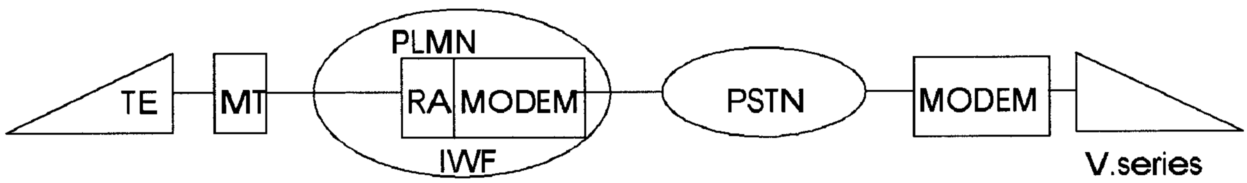


Figure 1
PLMN PSTN interworking for circuit switched calls

9.2.1 Network interworking mobile originated

9.2.1.1 Selection of interworking function

The interworking function will need to negotiate with the user to establish the appropriate modem selection e.g. data rate, modulation scheme, etc. In addition, it will also be required to convert the signalling format, from a combination of out of band and in band, to that suitable for controlling the modem and the autocalling line procedure function where applicable. In the following modem selection procedures it is assumed that the interworking function will be contained within the PLMN, but that a common selection scheme will be appropriate either to a centralised modem resource, or to modems associated with each MSC. The major difference being the routeing necessary internal to the PLMN:

For a data call originated by a circuit mode data terminal on the PLMN, the modem selection is done by using some appropriate elements in the call set-up message (bearer capability).

Utilising the bearer capability information field, having elements such as: Information transfer rate, User rate, Information transfer mode, and signalling and information access protocol etc., it is possible to uniquely identify a modem type. This field however means that a user is only able to select the same modem type and speed, from the modem pool, as his terminal is utilising as the DTE/DCE interface at the mobile station e.g. V.22.

9.2.1.2 Parameters utilised for selecting modem and mode of operation

Parameter name	In band	Out of band (Call setup) (Call confirmation)	Out of band (Call setup Ack or proceeding)
Information transfer capability		Yes	
Info transfer rate		Yes	
Structure		Yes	
Configuration		Yes	
Establishment		Yes	
User rate	Yes	Yes	
Intermediate rate		Yes	
Sync/Async		Yes	
Start/stopp bits		Yes	
Data bits		Yes	
Parity		Yes	
DTE/DCE interface		Yes	
Network independent clock	Yes	Yes	
Flow control	Yes 2)	Yes	
Negotiation	Yes	Yes	
Progress indicator			Yes
Protocol identification		Yes	
Transparent/Non-transparent service		Yes 1)	
Half/Full rate		Yes 1)	

TABLE 5/09.07
Parameters utilised for selecting modem and mode of operation

1 Not included in ISDN recommendation CCITT Q.931.
2 Only for non-transparent services to ISDN

9.2.1.3 Modem Selection

In general terms the indication of the bearer capability parameter "Information transfer Capability" will be utilised in the call set-up message to determine when the modem should be selected in the call. For the attribute value "3.1 kHz audio Ex PLMN", the modem will be selected immediately. The autocalling procedure according to V.25 bis will then be carried out supported by the respective signalling functions of the network and the corresponding modem functions (1300 Hz tone sending, 2100 Hz tone recognition, etc.) For the attribute values "Alternate speech/unrestricted digital" (if speech is selected as the first service) and "Speech followed by unrestricted digital" then the modem is made available but not selected until the subscriber indicates the change of service request (see section 9.3).

Note: A modem may also be selected by the PLMN, if the MS indicates a bearer capability information element with the attribute values "unrestricted digital information" and "no rate adaptation", and a modem type is available.

9.2.1.4 DTE/Modem interface (Filtering)

The DTEs taken into account for the PLMN at the MS side conform to CCITT's DTE/Modem interface specifications, which assume basically an error-free environment, i.e.

* limited distance, point-to-point local interconnection of the interface circuits for data and status

* steady state signalling.

The envisaged use of these DTE's in the PLMN environment leads to the exposure of these "interconnections" to the PLMN Radio Channel. To assure proper operation even under these conditions appropriate measures have to be taken. In the "non-transparent case" the RLP satisfies the requirement for both data and status lines. In the "transparent" case, the

* data line aspects have to be dealt with end-to-end between the users, while

* status line aspects are of concern to the network which are dealt with in the following.

The use of the channel control information for the remote control of the DTE/Modem control interchange-circuits between the MS and the IWF (the conveyance of which is supported by the rate adaptation scheme adopted for PLMN application) requires alignment to the particular transmission occurrences in the traffic channel to be taken into account within the PLMN. In principle this can be best achieved by

- relying only on the PLMN outband signalling as far as connection control is concerned

- eliminating the dependence upon the transmission of channel control information via the radio link.

Support for this strategy is given to a certain extent by the confinement of PLMN data connection to

- full duplex operation

- switched service (demand access)

- mapping of connection-control relevant conditions of the DTE/DCE control interchange-circuits to/from outband PLMN signalling according to Rec. GSM 04.08.

- flow control supported only in non-transparent mode
- support of connections with the same user data rate only (no TA end-to-end flow control in case of transparent mode).

The only DTE/Modem control interchange-circuit conditions, which actually are not covered by the above confinements, are the indications of readiness for data transmission, i.e. CT106/109 in case of V.-series interface and I-circuit in case of X.-series interface. As the effect of a conditions change of the aforementioned DTE/Modem interchange-circuits depends on the

- phase within the course of the connection
- direction of change (ON-OFF or OFF-ON)

The required precaution to be applied (Filtering) must be determined individually in view of

- function deduced from the change
- resilience of the connection needed
- error condition possibly invoked due to a delay in performing the condition change of the control interchange circuit
- potential loss of performance in connection usage.

The details of the filtering function are laid down in GSM 07-series Recommendations.

9.2.1.5 <u>Subscription checking for mobile originated PSTN terminated call</u>

This is handled in the same way as for ISDN terminated calls. See section 10.2.1

9.2.2 <u>Network Interworking Mobile terminated PSTN Originated</u>

This section describes the interworking of calls where the calling subscriber cannot generate or communicate a bearer capability to a PLMN (gateway MSC/interrogating node) because of lack of ISDN signalling capability.

Two methods of allocating MS International ISDN Numbers (MSISDNs) are allowed: Firstly, a separate MSISDN may be allocated for each service, or service option, which a subscriber uses for incoming calls; or, alternatively, a single number, applicable for all incoming calls is used.

a) Multiple MSISDNs are used ("The Multi-numbering Scheme"). See figure 2.

   In this scheme, the HPLMN will allocate a number of MSISDNs to a subscriber and associate with each of these numbers some interworking information ("IWI"). This IWI may be one or two complete Bearer Capability (BC) code(s) (coded as per GSM Recommendation 04.08) plus HLC or subset (eg. bearer service code plus modem type). In either case, when the HLR receives an interrogation relating to an incoming call (ie. the MAP "Send Routing Information" procedure), it requests a roaming number from the VLR. This request will contain the GSM BC(s) reflecting the service associated with the called MSISDN, ie. the IWI is translated to GSM BC(s) and passed to the VLR.

At the VLR, when the incoming call arrives, the BC associated with the MSRN is retrieved from the VLR and sent to the MS at call set-up.

On receipt of a Set-up message containing a BC, the MS will analyse the contents to decide whether the service can be supported (with negotiation as necessary, see below) and the call will be accepted or rejected as appropriate .

Certain of the parameters of a received BC can be negotiated. These are: Connection Element (Transparent\non-transparent), Radio Channel Requirements (Half\Full Rate), number of data bits, number of stop bits and parity. In addition an user information layer 2 protocol element may be returned, if no previously indicated. This negotiation takes place by means of the MS reflecting back to the MSC a complete bearer capability code in the call confirm message, with the relevant parameters changed. If this does not take place (ie. if there is no BC present in the call confirmed message), than the MSC will assume that the values originally transmitted to the MS are accepted.

In the case where the Home network does not allocate one or two complete BC(s) to each MSISDN then the MS must have prior knowledge to allow the correct negotiation of the negotiable parameters (although, of course, relevant non-negotiable parameters should be associated with the MSISDN). In the case where a complete BC is allocated, the user is unlikely to negotiate since the calling party should match the stated parameters. In either case negotiation may or may not take place at the user's discretion. The handling of these parameters by the home network is invisible to the visited network.

b) A Single MSISDNs is used ("The Single-numbering Scheme"). See figure 3.

In the single-numbering scheme, the HPLMN will allocate one MSISDN to a subscriber, applicable to all services.

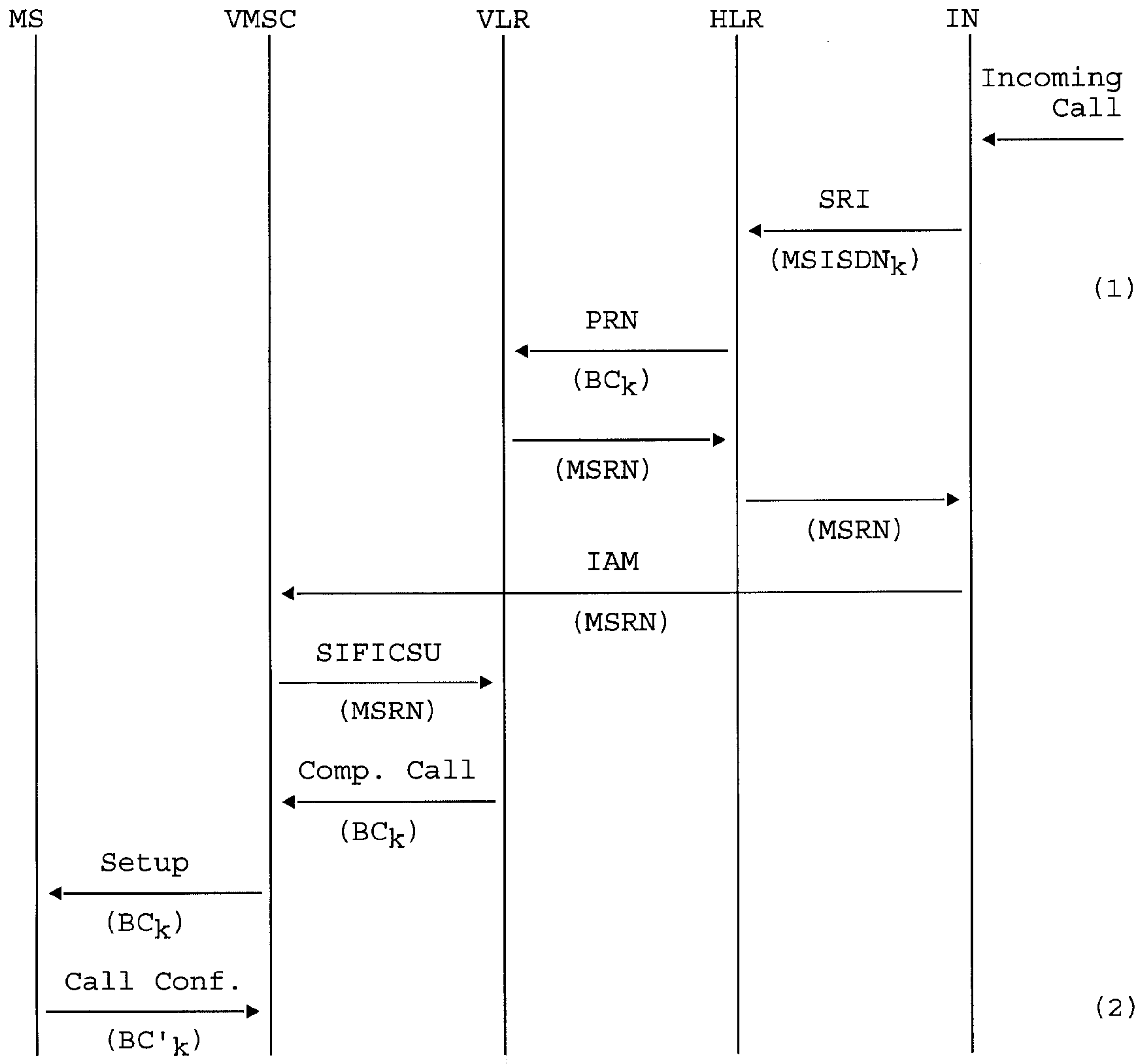
In this case, when the HLR receives an interrogation relating to an incoming call (ie. the MAP "Send Routing Information" procedure), the request to the VLR for a roaming number will not contain a GSM BC.

At the VLR, when the incoming call arrives, there is no BC associated with the MSRN and so the call set-up to the mobile will not contain the GSM BC element.

In this case, the MS will return a complete GSM BC in the Call Confirmed message, indicating the service required by the mobile subscriber. The VMSC will analyse this BC and if the requested BC can be supported the call is established, otherwise the call will be released.

Figure 2

Mobile terminated, PSTN originated call; HLR uses multiple MSISDN numbers with corresponding BCs.



Notes:

(1) The HLR translates the received MSISDN called adress ($MSISDN_k$) into the relevant bearer capability information (BC_k).

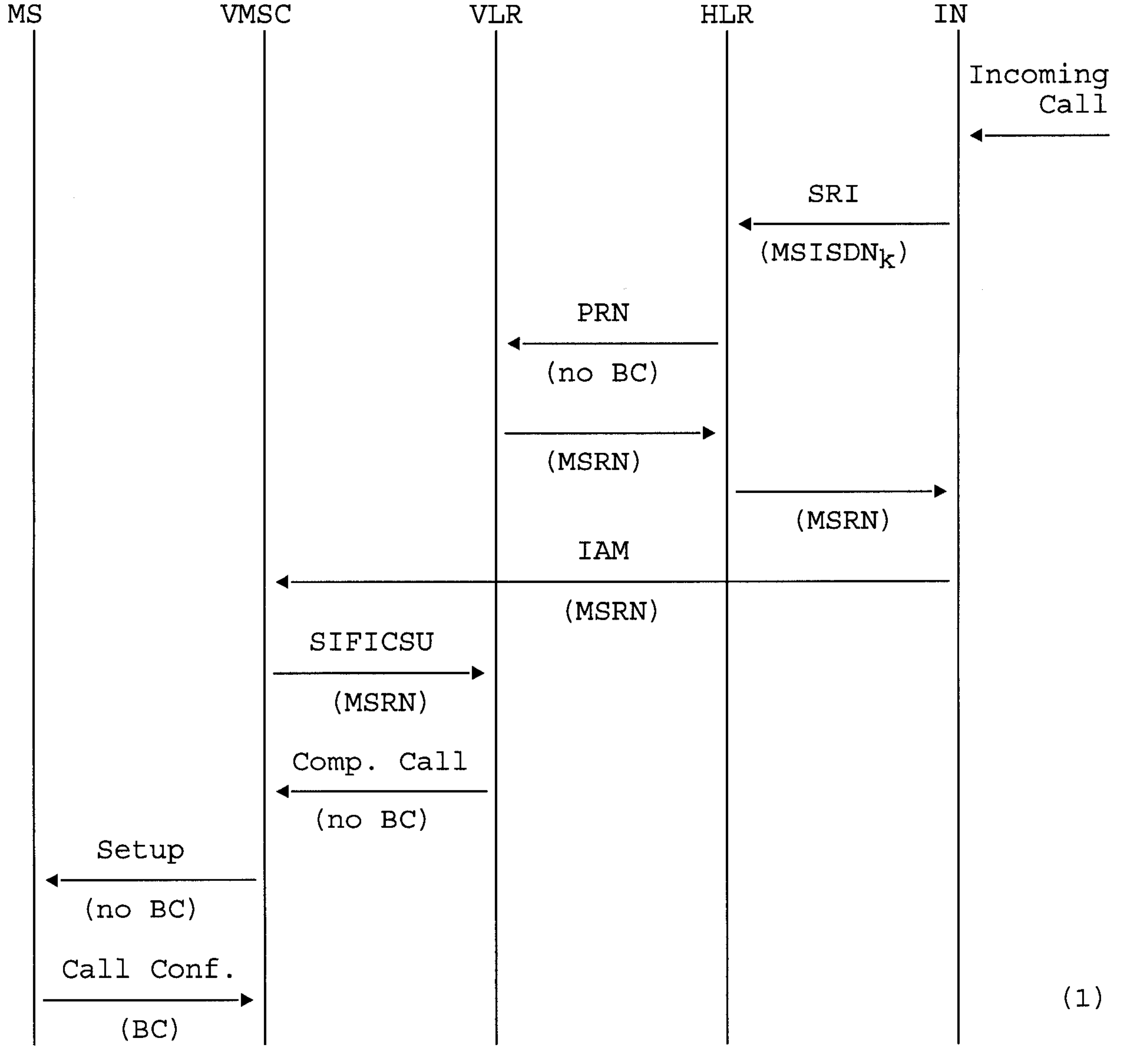
(2) In the "Call Confirm" message, the MS may modify some parameters of the BC. See 9.2.2.

Abbr.:

SRI - Send Routing Information
PRN - Provide Roaming Number
MSRN - Mobile Station Roaming Number
IAM - Initial Address Message
SIFICSU - Send Information For Incoming Call Set Up

Figure 3

Mobile terminated, PSTN originated call; HLR uses single MSISDN numbers (no corresponding BC stored). Per call MSRN allocation.



Note: (1) This BC is derived from information stored in the Mobile Station, according to its configuration.

Abbreviations: see figure 2.

9.2.3 Transparent service support (see Recommendation 03.10)

Recommendation 08.20 identifies the rate adaptation scheme to be utilised on the BS to MSC link. The transcoding function will re-generate the 64kbit/s rate adapted format utilising the 8 and 16kbit/s intermediate data rates. The MSC to IWF will utilise the same rate adaptation scheme as that indicated in recommendation 08.20, but further adapted to 64kbit/s.

For the transparent service support the IWF will select the modem and speed based on the information contained in the call set-up message. Where the modem type indicated is one of the multi-speed versions, e.g. V.32, then the IWF will restrict the modem to the speed indicated in the call set-up, i.e. will inhibit the modem from changing speed, irrespective of the conditions, error rate, encountered on the PSTN link. This scenario is also applicable for the use of "autobauding" modems, in that only the specifically requested modem type and speed will be selected at the IWF.

9.2.3.1 MSC to IWF rate adaptation scheme

This link consists of a 64kbit/s channel with the information, both user data and in band parameter information (where provided) rate adapted in conformance to Rec 08.20.

9.2.3.2 Rate adaptation process in IWF

This process is a reverse of that provided in the Terminal Adaptation function of the MS. Recommendation 04.21 refers to the rate adaptation mechanism to be provided.

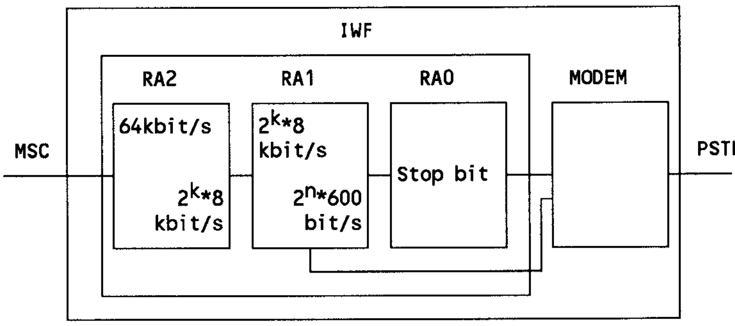


Figure 4
Rate adaptation schematic

9.2.3.3 Mapping of signalling MS/IWF to modem interface requirements

This process also is a reverse of the function provided in the TA recommendation for the mapping of DTE/DCE signalling information to Dm channel and in band signalling information. Recommendations 07.02, and 07.03 refer.

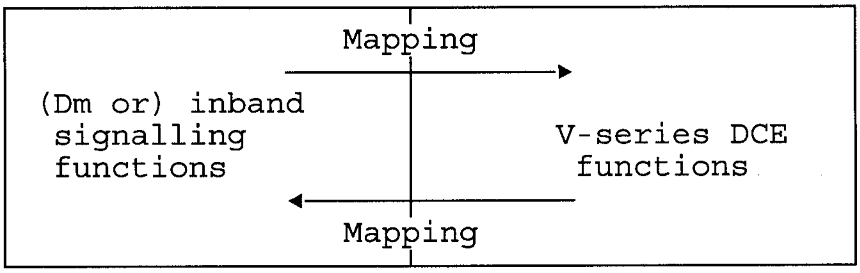


Figure 5
Signalling mapping schematic

In general it is not required for the modem in the IWF to support a "remote looping" request from a modem in the PSTN. In addition the invocation of a "remote looping" request from the mobile subscriber to a modem in the PSTN need not be supported. Specific test loops for mobile subscribers to contact may be provided at the network operators discretion.

9.2.3.4 Establishment of end-to-end terminal synchronizations

Prior to exposing the traffic channel of a PLMN connection to transmission of user data, the controlling entities of the connection have to assure of the availability of the traffic channel. This is done by a so called synchronizations process:

- starting on the indication of "physical connection established" resulting from the PLMN-inherent outband signalling procedure

- ending by indicating the successful execution of this process to the controlling entity, which then takes care of the further use of the inband information (data, status).

Network interworking within an IWF is concerned with the terminating side (to the MS) and the transit side (to the fixed network) of a connection. Both sides have to be treated individually related to the synchronizations process.

With respect to the terminating side the procedure is as follows:

- sending of synchronizations pattern 1/OFF (all data bits"1"/all status bits "OFF") to the MS using the RA1/RA2 rate adaptation function

- searching for detection of the synchronizations pattern 1/OFF from the MS.
- holding the modem interchange circuits (with the exception of CT108) in the OFF condition until timer T (see below) expires, when they are switched to ON.

When the 1/OFF from the MS has been recognised as a steady state, the IWF continues sending the synchronizations pattern 1/OFF to the MS unless a timer T (= 500 ms) expires. From this time the information on CT106 and CT109 from the MS are directly mapped to the respective sending lines.

Mobile Originated

At the start of timer T, i.e. on receipt of the synchronizations pattern from the MS, circuit 108 to the selected modem associated with the connection will be switched from the "OFF" to "ON" condition, thus initiating the auto calling sequence.

Mobile Terminated

At the start of timer T, i.e. on receipt of the synchronizations pattern from the MS, circuit 108 to the selected modem associated with the connection will be switched from the "OFF" to "ON" condition.

It should be noted that in a GSM-PLMN V.-series and X.-series interfaces are only supported in full duplex mode. Thus the call control phase can be mapped almost completely to the signalling procedure (the S-bits during the call control phase are irrelevant). However, the "ready for data" condition (i.e. CT106/109, in case of V.-series interface, and I-circuit, in case of X.-series interface) is derived directly from the traffic channel (see also filtering of channel control information).

9.2.3.5 Network Independent Clocking (NIC)

Within the GSM network the coding of the values for bits associated with NIC is specified in GSM recommendations 04.21/08.20. In the forward (transmitting) direction the multiframes shall be coded in exact accordance with that specified in those recommandations. Bit E6 is set to "1" in alternate modified v.110 frames at the transmitter. However, the use of this bit at the receiver for monitoring frame Synchronisation, or any other purpose, is not specified and is left to the discretion of the inplementor.

A "perfect linear block Code" is used in C1-C5, whose error correction properties may be utilised in the receiver, in order to ensure reliable operation of NIC.

The NIC sending function has to recognise when the difference between the applicable clock speed of the GSM network and the interface speed generates a positive or negative whole bit requirement. When this positive or negative condition occurs, the NIC codewords specified in recommendation GSM 04.21 are used to transport this condition to the receiving NIC function. Transmission of the codeword shall clear the positive or negative condition related to that codeword at the sending function. The sending function shall not send more than one positive or negative compensation within a contiguous period of time corresponding to 10000 user data bits minus the number of user data bits necessary to make up an even number of V.110 frames between compensations (NIC compensation is coded in two V.110 frames). This results from the requirements to compensate for maximum clock differences of ±100 parts per million. If the receiving function receives NIC compensations more often than a contiguous period of time corresponding to 10000 user data bits, there is no guarantee that data will not be lost.

The NIC receiving function has to provide the capability to support the compensation requirements of the sending function. This compensation is managed by manipulating the clock speed of the interface, within the standard constraints of that interface.

Overall, the compensation functions have to be capable of managing clock tolerances of ±100 parts per million.

Action on loss of synchronisation

If five consecutive NIC multiframes have incorrect framing bit values in E7, the receiver shall stop applying clocking compensation to the received data. Resynchronisation will be attempted and compensation will resume when synchronisation is achieved.

9.2.4 Non-transparent service support (see Rec. 03.10)

Recommendation 08.20 identifies the rate adaptation scheme to be utilised on the BS-MSC link, as for the transparent case.

9.2.4.1 MSC-IWF Rate adaptation scheme

This will be the same as for the transparent case.

9.2.4.2 Protocol layer structure in the IWF

Recommendation 03.10 identifies the protocol layer structures for the non-transparent case, the physical layer to the PSTN is provided by means of a modem.

9.2.4.3 Re-constitution of user data

Recommendation 04.22 refers to the frame of user data in the radio link protocol. The layer 2 relay function in the MS (identified in rec.03.10) contains the mechanism for the stripping of the start and stop bits, for asynchronous applications, and synchronizations of the user data within the RLP frame. The L2R function within the IWF performs the reverse of this role, for transmitted data by re-inserting the start and stop bits and the stripping of received data.

9.2.4.4 Layer 2 relay functionality

Specific functionality is required of the L2R dependant upon the service which is being requested to be supported. The selection of the appropriate L2R function will be determined by the IWF on the basis of the bearer capability information signalled in either the call set-up request, or call confirmation messages. The prime information element being transparent or non transparent service indication. In addition the particular L2R function will be selected on the basis of the users layer 2 indicated e.g. X.25, IA5, etc.

The specific interaction between the L2R function and the RLP function and the L2R frame structure will be the same as that detailed in the Annex to the appropriate 07 series recommendation.

9.2.4.5 In band signalling mapping flow control

This entails the L2R function providing the means of controlling and responding to flow control functions of the modem plus any synchronizations requirements related to flow control.

Normally, the IWF will utilise in band (X-on/X-off) flow control, although out of band (RTS/CTS) flow control will be provided in so far as V.42 procedures are able to support it.

Whichever flow control is provided, the L2R function must

(a) provide immediate indication of flow control to the fixed network on receipt of flow control request from the MS.

and/or

(b) provide immediate indication of flow control to the MS on receipt of flow control request from the fixed network i.e. in the next available L2R status octet to be transmitted.

Where in band (X-on/X-off) flow control is in use, then the X-on/X-off characters will not be passed across the radio interface.

9.2.4.5.1 Conditions requiring flow control towards the fixed network

The L2R function will initiate flow control in the following circumstances:

1) The transmit buffer reaches a preset threshold.

2) The L2R function receives an explicit "flow control active" indication.

On removal of buffer congestion or receipt of L2R "flow control inactive" the flow control will be removed.

9.2.4.5.2 Conditions requiring flow control towards the MS

The L2R function will transmit to the MS an explicit "flow control active indication" in the following circumstances:

1) If the receive buffer from the radio side reaches a preset threshold.

2) If a flow control indication is received from the fixed network customer. On receipt of this flow control indication, transmission of data from the receive buffers towards the fixed network terminal is halted.

On removal of the buffer congestion or fixed network flow control indication, the L2R function will send a "flow control inactive" indication towards the MS. In addition, for the fixed network indication, transmission of data from the receive buffers will be restarted.

9.2.4.6 Data buffers

9.2.4.6.1 Transmit buffers (towards MS)

Incoming data from the fixed network customer shall be buffered such that if the IWF is unable to transfer data over the radio path the data is not lost.

The buffer shall be capable of holding [16-32 kbits]. When the buffer is half full flow control towards the fixed network shall be initiated as per paragraph 9.2.4.5.1.

9.2.4.6.2 Receive buffers (from MS)

Incoming data from the MS is buffered such that if the fixed network terminal is unable to accept the data then it is not lost.

The buffer shall be capable of holding [16-32 kbits] of data. When the buffer becomes half full, the L2R function will send a "flow control active" indication towards the MS, as per paragraph 9.2.4.5.2.

9.2.4.7 Transportation of the "Break" condition

The "Break" condition must be recognised by the L2R function and passed immediately to the MS. The L2R function will generate a break condition towards the fixed network on receipt of a break indication from the MS. Information in the buffers will be optionally discarded. (Further study is required regarding provision of a non-destructive BREAK condition).

9.2.4.8 In band signalling mapping modem status information

Status information from the modem will be carried by the L2R function to/from the L2R function in the terminal adaptation function. The IWF is not intended to utilise this information for any purpose. The use of "Data carrier detect" or "clear to send" by the terminal adaptation function to determine PSTN link establishment or failure is not utilised by the IWF; e.g. call clearing, in event of line failure, will be generated normally by the MS not the IWF.

9.2.4.9 Establishment of end-to-end terminal synchronizations

Prior to exposing the traffic channel of a PLMN connection to transmission of user data, the controlling entities of the connection have to assure of the availability of the traffic channel. This is done by a so called synchronization process

- starting on the indication of "physical connection established" resulting from the PLMN-inherent outband signalling procedure

- ending by indicating the successful execution of this process to the controlling entity, which then takes care of the further use of the inband information (data, status).

Network interworking within an IWF is concerned with the terminating side (to the MS) and the transit side (to the fixed network) of a connection. Both sides have to be treated individually related to the synchronizations process.

With respect to the terminating side the procedure is as follows:

- waiting for the RLP link establishment by the MT (in addition the IWF may initiate the RLP establishment).

When the RLP has been established, CT108 will be turned ON to enable the autocalling/autoanswering function of the selected modem. From this time the information from/to the RLP including status changes will be mapped by the L2R entity applicable to the particular bearer capability.

It should be noted that in a GSM-PLMN V.-series and X.-series interfaces are only supported in full duplex mode. Thus the call control phase can be mapped almost completely to the signalling procedure (the S-bits during the call control phase are irrelevant). However, the "ready for data" condition (i.e. CT106/109, in case of V.-series interface, and I-circuit, in case of X.-series interface) is derived directly from the traffic channel (see also filtering of channel control information).

The synchronizations process described above applies to both the transparent and the non-transparent mode of transmission. However, the exchange and detection of the synchronizations pattern itself is only essential for the transparent mode. Otherwise the RLP automatically assures the availability of the traffic channel".

9.3 Interworking Alternate Speech Data Calls

9.3.1 Alternate Speech Data Bearer Interworking

9.3.1.1 General

The procedure for alternate speech unrestricted digital bearer service is invoked at the call set-up phase. This service is invoked by indication of the "Information transfer capability", alternate speech/unrestricted digital. In addition to this, the call set-up message should include the use of repeated bearer capability messages, one indicating speech and the other indicating the specific data user rate etc., as for normal data calls. The bearer capability first indicated i.e. speech or digital unrestricted determines the first selection required of the network by the subscriber. Optional lower layer and higher layer capabilities may also be included. The IWF will perform both compatibility checking and subscription checking on both sets of capabilities as for normal data calls. If this check fails then the call will be rejected.

The speech capability will utilise normal telephony teleservice interworking requirements and mobile network capabilities. The unrestricted digital capability will utilise the appropriate data interworking capability and may use either the transparent or non-transparent mobile network capability.

9.3.1.2 Mobile originated PSTN terminated calls

The call is set up in the normal manner and handled by the IWF as indicated in the general section. The speech section of the call, when invoked, is handled by the transcoder and IWF as for normal telephony teleservice call, this includes any requirements for echo cancellers etc. as indicated in paragraph 9.1. The data element of the call, when invoked utilises an appropriate modem, selected from the information contained in the bearer capability information. The network shall provide, for service and operational reasons, a rapid and reliable changeover of capability upon request from the user. This changeover may involve the disabling, by-passing or introduction of particular network functions (e.g. speech coder, modem etc.). This changeover is initiated on the receipt of a signalling message from the mobile subscriber.

9.3.1.3 PSTN originated mobile terminated calls

The call set up and request for this particular service is performed in a similar manner to that indicated in paragraph 9.2 for normal PSTN originated calls. In addition, taking into account the BC format as described in the general section (9.3.1.1), when multiple MSISDNs are used ("Multi-numbering scheme") to allow auto answering mode for the data phase (i.e. the call starts automatically with the data phase), the MS can reflect back to MSC the dual Bearer Capability in the Call Confirm message with the BC elements interchanged to those in the original Call Set-up message (i.e. data element first). In all other aspects it is handled as indicated for mobile originated.

9.3.2 Speech followed by data interworking

9.3.2.1 General

The set up and selection of interworking function for this service is the same as that indicated for the alternate speech digital service. The only difference in this service is that speech will always be the first bearer capability selection and once the changeover command is received from the mobile station then all network resources associated with the handling of the speech call may be released for reallocation to other calls, i.e. they will not be required again in the handling of this call. Both mobile originated and terminated are dealt with as detailed in section 9.3.1.2 and 9.3.1.3.

10. Interworking to the ISDN

The scope of this section is to describe the handling of the content of the Information Elements where "content" is understood to be the value of the parameter fields of the Information Elements, namely BC-IE, HLC and LLC, after the length indicator. For the transport of these Information Elements within the PLMN refer to GSM 09.02.

10.1 Speech Calls

Since at the interworking point the transcoder provides for A law PCM at 64kbit/s, no particular interworking is required. It is anticipated that the ISDN Teleservice Telephony would be used. Transmission aspects are covered in GSM recommendation 03.50. Any further requirements are a national matter.

10.2 Data Calls

In this case it is assumed that the ISDN bearer service 3.1 kHz audio shall only be interworked by means of a modem pool in the PLMN. If a network operator provides this facility, then the IWF operation will be similar to that described for interworking to the PSTN.

10.2.1 Network interworking mobile originated

Lower layer compatibility checking of the mobile originated call is carried out by the IWF to determine the appropriate bearer service selection in the ISDN. This will entail the IWF in modifying some parts of the bearer capability information field. If it is not possible for the IWF to provide a bearer service match, then the IWF shall fail the call and indicate the reason to the user.

As well as compatibility checking, the VLR should check the subscribers subscription parameters against information received from the VMSC in the "send information for outgoing call Set-up" message. The mapping from call parameters to service codes is contained in annex 1 [to be provided]. This ensures that the calling mobile does subscribe to the requested service.

10.2.1.1 Circuit switched calls

Where the bearer capability information indicates that the call is a circuit switched unrestricted digital call, then the IWF shall select the appropriate rate adapted ISDN bearer service.

The selection of the IWF will be by means of the bearer capability information within the call set up message. The mobile subscriber shall be able to select the unrestricted digital capability, which the IWF will map to the same capability in the ISDN call set up message. If an interworking point is encountered within the ISDN which does not support this service request, then a cause failure message indicating network unable to support service requested will be returned to the PLMN which will then pass this to the mobile subscriber. This will be used at the MS to clear the call. It will then be possible for the mobile subscriber to initiate a new call request this time indicating the transfer capability "3.1 kHz Ex PLMN" plus other attributes such as user rate, etc. The IWF plus network will then interpret this requirement as:

1) A connection type within the mobile network of unrestricted digital with the appropriate user rate, intermediate rate adaptation, etc. selected.

2) The appropriate modem is selected at the IWF (determined from the bearer capability information).

3) The 3.1 kHz audio bearer service is selected in the ISDN.

This process will then enable the call to progress through the interworking point in the ISDN.

Note: The network may also select the ISDN Bearer Service "3.1 kHz audio" if the MS indicates "unrestricted digital information" and "no rate adaptation", and a modem type is available.

10.2.1.2 Packet calls

The mobile network offers only Bm channel access for the packet mode service. The ISDN offers both B and D channel access for the packet mode service. The interworking of mobile packet calls is described in Recommendation GSM 09.06.

10.2.2 Network interworking mobile terminated

This section describes the interworking of calls where the calling subscriber can communicate a bearer capability to a PLMN (gateway MSC/interrogating node) i.e. by means of ISDN signalling.

The GMSC has to perform a mapping of the received Basic Service Information for the transport to the HLR, for details of this transport refer to GSM 09.02.

Compatibility checking of the lower layers of the ISDN originated call is carried out by the IWF to determine the appropriate bearer service selection in the PLMN. This will entail the IWF in modifying some parts of the bearer capability information field. If it is not possible for the IWF to provide a bearer service match, then, providing the user has indicated that negotiation is possible, the IWF may negotiate with the user to change the bearer service requested. If negotiation is not possible then the IWF shall fail the call and indicate the reason to the user.

As well as compatibility checking, the HLR should check the subscribers subscription parameters against the call parameters received in the "send routing information" message from the IN. This ensures that the called mobile does subscribe to the requested service.

For ISDN originated calls it will not be possible to signal mobile specific requirements e.g. transparent/non transparent, full/half rate channel. In this case for incoming calls the IWF shall select a default setting appropriate to the PLMN's network capabilities. In general it will be beneficial, where a network supports both full and half rate channels and transparent/non transparent capabilities, to indicate so in the appropriate GSM BC field of GSM 04.08. The mobile subscriber has the option to indicate in the call confirmation message a change to this default setting. The appropriate IWF shall be selected on the basis of this requirement.

At call Set-up, the interrogating node passes in the "send routing information" to the HLR, the ISDN BC[1] received in the initial address message. The coding of these parameter values must comply with recommendation T/S 46-30.

According to the ISDN BC received, the HLR applies one of the following alternatives:

1. An ISDN BC is received with the information Transfer Capability field set to "3.1 kHz audio" but without any associated modemtype or HLC indication of group 3 facsimile. Two cases have to be considered:

    a) The called MSISDN has a corresponding BC stored in the HLR (see option a of 9.2.2); then the service attached to this number in the HLR tables is applicable and the corresponding BC is passed to the VLR in "provide roaming number" See figure 6.

    b) The called MSISDN has no corresponding BC stored in the HLR (see option b in 9.2.2); In this case no BC is passed to the VLR in the "provide roaming number" message.

2. The ITC field in the BC received is not 3.1 kHz audio, or if 3.1 kHz audio, a modem type or facsimile group 3 is indicated in the HLC. This received BC is then considered applicable regardless of the kind of MSISDN received (BC associated or not) and the equivalent GSM BC is sent to the VLR. See figure 7.

At the MSC, when the incoming call arrives, the BC associated with the MSRN is retrieved from the VLR and sent in general to the MS at call set-up. In case there is no BC stored in the VLR, the call set-up message to the MS will not contain any BC-IE. In particular, however the following rules apply:

1. If the initial Address Message (IAM) contains no BC and there is no GSM BC retrieved from the VLR, the call is handled as section 9.2.2 case b. (This case may arise if the HLR does not use the "Provide Roaming Number" but uses MSRN from location updating.)

2. If there is no BC in the IAM and a GSM BC has been attached to the MSRN when it was allocated, the retrieved BC applies.

3. If there is a BC in the IAM with the ITC field set to 3.1 kHz" but without any associated modem type or indication of facsimile group 3, the BC retrieved from the VLR is considered as applicable when it exists. If no BC is retrieved from the VLR, the call is handled as in section 9.2.2.

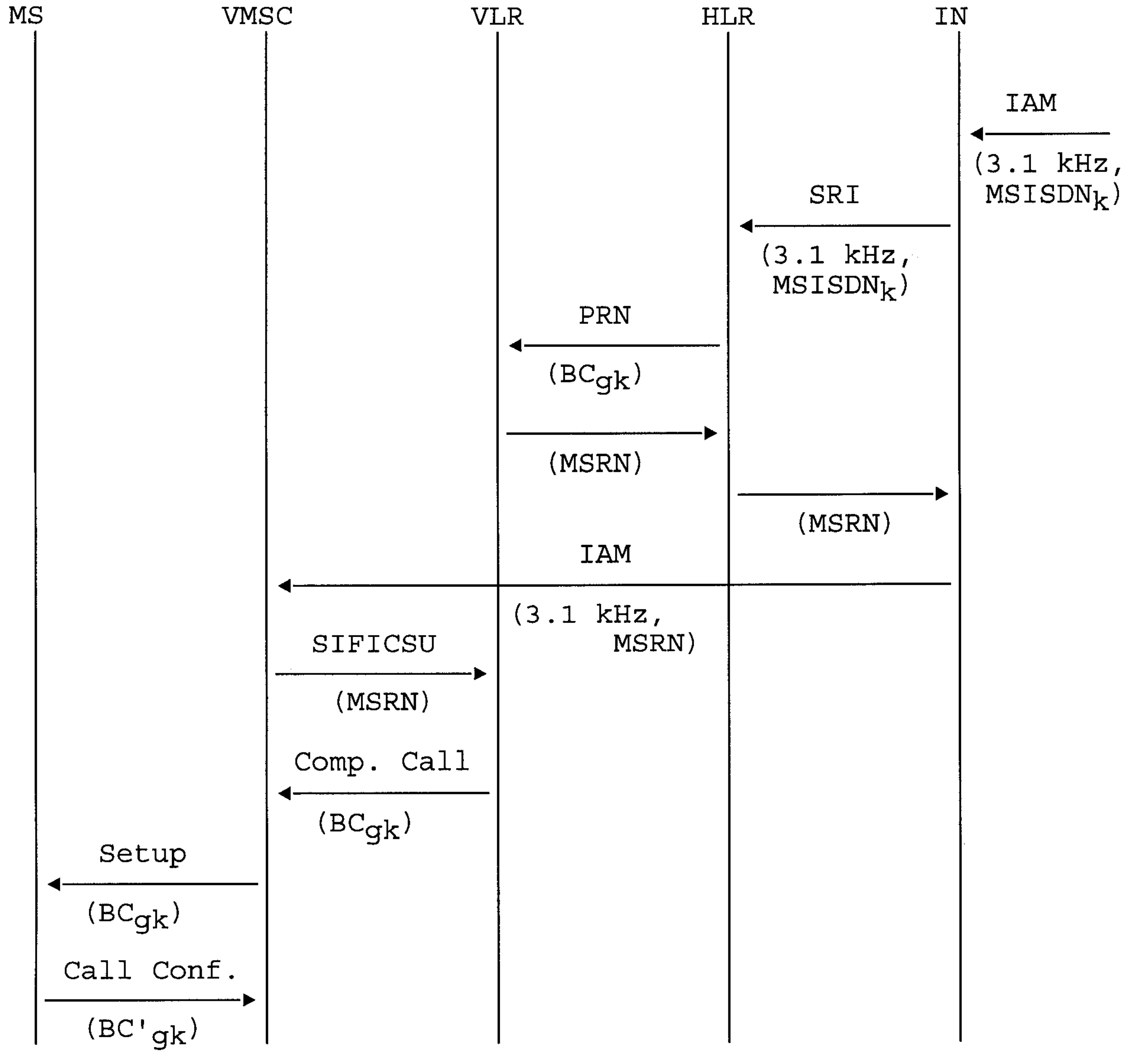
4. If the BC received in the IAM has the ITC field set to the value different from 3.1 kHz audio or if 3.1 kHz audio, a modem type or facsimile group 3 is indicated, this BC is applicable regardless of what has been retrieved from the VLR.

The mapping between GSM and ISDN BCs is shown in table 6/09.07.

1) Editorial note: In addition, ISDN HLC+LLC will also be sent from the IN to the HLR.

Figure 6

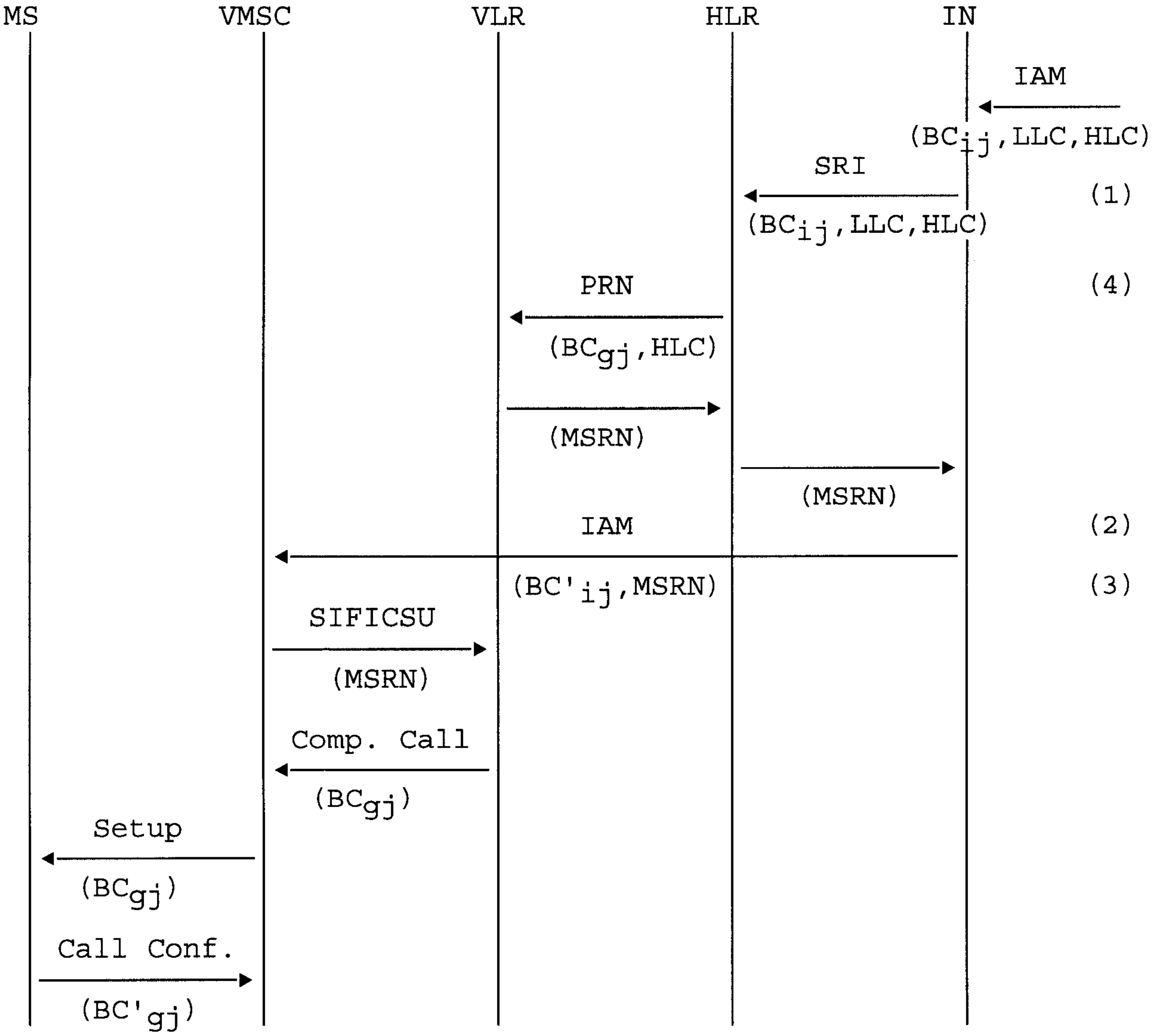
Mobile terminated, ISDN originated call; HLR stores BC against MSISDN number. Information Transfer Capability = 3.1 kHz audio.



Abbreviations: see figure 2.

Figure 7

Mobile terminated, ISDN originated call; HLR uses multiple or single number tables. Information Transfer Capability = 3.1 kHz audio.



Notes:

(1) BC_{ij} denotes ISDN (T/S 46-30) BC*. For details on how the BC_{ij}, LLC and HLC are transported refer to GSM 09.02.

(2) Assumes signalling capabilities permit the transfer of BC between IN and VMSC. If this is not the case, the VLR uses the stored BC.

(3) BC'_{ij} denotes BC_{ij} as maybe modified by intervening networks.

(4) BC_{gj} is the corresponding GSM BC. For details on how the BC_{gj} and HLC are transported refer to GSM 09.02.

* Editorial note: In addition, ISDN HLC and LLC.

Abbreviations: see figure 2.

The following tables (6A + 6B) show that presently only the ISDN BC is used for mapping (exceptions are indicated). LLC and HLC elements are passed unchanged where possible. In particular the LLC is neither used for mapping nor created by the network.

Table 6A-09.07:
Comparable setting of parameters in GSM 04.08 and T/S 46-30 (ETSI ISDN user to network signalling recommendation) Mobile Originated.

In the following table the comparision is drawn between parameters in the GSM call set up request message and that of the ISDN call set up request message. In some cases no comparable values are available and these will be marked as such. In these cases reference will need to be made to the table of network interworking in recommendation GSM 09.07 to identify the appropriate choice. In some cases it is not necessary to support a particular option, and in this case those parameters will be annotated appropriately.

Octet	GSM 04.08 parameter value	Octet	T/S 46-30 parameter value
1	Bearer Capability IEI	1	Bearer Capability IEI
2	Length of BC contents	2	Length of BC contents
3 #7..6	Radio channel requirement ▪ half rate channel ▪ full rate channel ▪ dual, half rate preferred ▪ dual, half rate preferred		No comparable value
3 #5	Coding Standard ▪ GSM standardized coding	3 #7..6	Coding standard ▪ CCITT standardized coding
3 #4	Transfer mode ▪ circuit mode ▪ packet mode	4 #7..6	Transfer mode ▪ circuit mode ▪ packet mode
3 #3..1	Information transfer capability ▪ speech ▪ unrestric. digital(note 2) ▪ 3.1 kHz audio ex PLMN ▪ speech followed by unrestricted digital ▪ faximile group 3 (note 1) ▪ not supported ▪ -- " -- ▪ -- " --	3 #5..1	Information transfer capability ▪ speech ▪ unrestricted digital ▪ 3.1 kHz audio ▪ see table of Service Interworking in GSM 09.07 ▪ -- " -- ▪ restricted digital inform. ▪ 7 kHz audio ▪ video
3a	Coding standard extension		No comparable value
4 #6..5	Structure ▪ SDU integrity ▪ unstructured	4a #7..5	Structure ▪ SDU integrity ▪ default

4 #4	Duplex mode ■ half duplex ■ full duplex	5d #7	Duplex mode ■ half duplex ■ full duplex
4 #3	Configuration ■ point to point	4a #4..3	Configuration ■ point to point
4 #1	Establishment ■ demand	4a #2..1	Establishment ■ demand
5 #5..4	Rate adaptation ■ no rate adaptation ■ V.110/X.30 rate adaptation ■ CCITT X.31 flag stuffing ■ No comparable value, however speech service interworking will indicate these requirements ■ Not supported ■ -- " -- ■ -- " --	5 #5..1	User information layer 1 protocol ■ no comparable value ■ CCITT standardized rate adaption V110/X.30 ■ CCITT standardized rate adaption X.31 flag stuff. ■ Recommendation G711 U law, Recommendation G711 A law, Recommendation G721 32 kbit/s ADPCM and I460 ■ Recommendation G722 and G725 7 kHz audio ■ Recommendation G7xx 384 kbit/s video ■ Non standard CCITT coding
5 #3..1	Signalling access protocol		No comparable field
6 #1	Synchronous / asynchronous ■ synchronous ■ asynchronous	5a #7	Synchronous / aynchronous ■ synchronous ■ asynchronous
6 #5..2	User info. layer 1 protocol ■ default layer 1 protocol	5 #5..1	User info. layer 1 protocol (see section under rate adaptation for 04.08 above)
6a #7	Number of stop bits ■ 1 bit ■ 2 bits ■ Not supported ■ -- " --	5c #7..6	Number of stop bits ■ 1 bit ■ 2 bits ■ not used ■ 1.5 bits
6a #6	Negotiation ■ In band neg. not possible ■ Not supported	5a #6	Negotiation ■ In band neg. not possible ■ In band neg. possible
6a #5	Number of data bits ■ 7 bits ■ 8 bits ■ No comparable value ■ Not supported	5c #5..4	Number of data bits excluding parity if present ■ 7 bits ■ 8 bits ■ not used ■ 5 bits

6a #4..1	User rate ■ 0.3 kbit/s ■ 1.2 kbit/s ■ 2.4 kbit/s ■ 4.8 kbit/s ■ 9.6 kbit/s ■ 12 kbit/s ■ 1.2 kbit/s / 75 bit/s ■ Not supported	5a #5..1	User rate ■ 0.3 kbit/s ■ 1.2 kbit/s ■ 2.4 kbit/s ■ 4.8 kbit/s ■ 9.6 kbit/s ■ 12 kbit/s ■ 1.2 kbit/s / 75 bit/s ■ rate is indicated by Ebits as specified in rec I460 ■ 0.6 kbit/s ■ 3.6 kbit/s ■ 7.2 kbit/s ■ 8 kbit/s ■ 14.4 kbit/s ■ 16 kbit/s ■ 19.2 kbit/s ■ 32 kbit/s ■ 48 kbit/s ■ 56 kbit/s ■ 64 kbit/s ■ 0.1345 kbit/s ■ 0.1 kbit/s ■ 75 bit/s / 1.2 kbit/s ■ 0.110 kbit/s ■ 0.115 kbit/s ■ 0.2 kbit/s
6b #7..6	Intermediate rate ■ not used ■ 4 kbit/s ■ 8 kbit/s ■ 16 kbit/s ■ Not supported	5b #7..6	Intermediate rate ■ not used ■ 8 kbit/s ■ 8 kbit/s ■ 16 kbit/s (note 3) ■ 32 kbit/s
6b #5	NIC on Tx ■ does not require ■ requires	5b #5	NIC on Tx ■ does not require ■ requires
6b #4	NIC on Rx ■ cannot accept ■ can accept	5b #4	NIC on Rx ■ cannot accept ■ can accept
6b #3..1	Parity information ■ odd ■ even ■ none ■ forced to 0 ■ forced to 1	5c #3..1	Parity information ■ odd ■ even ■ none ■ forced to 0 ■ forced to 1
6c #7..6	Connection element ■ transparent ■ non-transparent ■ both,transp. preferred ■ both,non-transp. preferred		No comparable field

6c #5..1	Modem type ■ none ■ V.21 ■ V.22 ■ V.22bis ■ V.23 ■ V.26ter ■ V.32 ■ modem for undef. interface ■ autobauding type 1 ■ Not supported	5d #6..1	Modem type ■ reserved ■ V.21 ■ V.22 ■ V.22bis ■ V.23 ■ V.26ter ■ V.32 ■ No comparable value ■ No comparable value ■ V.26 ■ V.26bis ■ V.27 ■ V.27bis ■ V.29 ■ V.35
7 #5..1	User info. layer 2 protocol ■ X.25 link level ■ IA5 ■ X.75 mod. lay 2 (teletex) ■ videotex profil 1 ■ facsimile group 3 ■ videotex profile 3 ■ Not supported	6	User info. layer 2 protocol ■ X.25 link level ■ no comparable value ■ -- " -- ■ -- " -- ■ -- " -- ■ -- " -- ■ Q921 (I441)

Other field values in the T/S 46-30 not supported in GSM recommendation 04.08 are:

Information transfer rate - In this case default 64 kbit/s selected.

Symmetry - In this case default bidirectional symmetric selected for all user data rates.

Flow control on transmission - This may be selected if V110 flow control procedure is provided.

Flow control on reception - This may be selected if V110 flow control procedure is provided.

V120 rate adaption octets - rate adaption header, multiple frame establishment, mode of operation, logical link identifier negotiation, assignor/ assignee, in band/out band negotiation - This protocol is not supported by GSM.

User information layer 3 protocol - X.25 packet layer selected when user information layer 2 protocol indicates X.25.

Note 1: In the case where GSM BC "Information Transfer Capability" indicates "Facsimile group 3" and only a single GSM BC is contained in the call set-up request then:

- If an HLC is not present, the network will insert a "Facsimile group 2/3" HLC.
- If an HLC element is present, the network will pass it through unmodified.

In the case where GSM BC "Information Transfer Capability" indicates "Facsimile group 3" and two GSM BCs are contained in the call set-up request, then the network takes no action regarding HLC generation or checking.

Note 2: In the following case the IWF may select an ISDN Bearer Service "3.1 kHz audio" with the TS46-30 parameter "Information Transfer Capability" set to "3.1 kHz audio":

- The GSM Bearer Capability parameter "Information Transfer Capability" indicates "unrestricted digital information"
  and
- Rate adaptation indicates "no rate adaptation"
  and
- a modem type is available.

Note 3: In the case of non transparent services, the value of the Intermediate Rate field of the ISDN Bearer Capability information element shall only depend on the value of the User Rate in the same information element. The correspondance is:

- Intermediate Rate = 16 kbit/s if the User Rate = 9.6 kbit/s
- Intermediate Rate = 8 kbit/s otherwise.

Table 6B-09.07:
Comparability and Mapping of bearer capability parameter values according to T/S 46-30 and GSM 04.08 within the HLR for a mobile terminated Call

Octet	T/S 46-30 parameter value	Octet	GSM 04.08 parameter value
1	Bearer Capability IEI	1	Bearer Capability IEI
2	Length of BC contents	2	Length of BC contents
	no comparable field	3 #7..6	Radio channel requirement (1) ■ half rate channel ■ full rate channel ■ both, half rate preferred ■ both, full rate preferred
3 #7..6	Coding standard ■ CCITT standardized coding	3 #5	Coding standard ■ GSM standardized coding
3 #5..1	Information transfer capability ■ speech ■ unrestricted digital ■ 3.1 kHz audio ■ no comparable value	3 #3..1	Information transfer capability ■ speech ■ unrestricted digital ■ 3.1 kHz audio ex PLMN (2) ■ facsimile group 3 (3)
4 #7..6	Transfer mode ■ circuit mode ■ packet mode	3 #4	Transfer mode ■ circuit mode ■ packet mode
4 #5..1	Information transfer rate ■ - ■ 64 kbit/s		no comparable field
4a #7..5	Structure ■ default (4) ■ 8 kHz integrity (5) ■ SDU integrity ■ unstructured	4 #6..5	Structure ■ no comparable value ■ no comparable value ■ SDU integrity ■ unstructured (*)
4a #4..3	Configuration ■ point-to-point	4 #3	Configuration ■ point-to-point
4a #2..1	Establishment ■ demand	4 #1	Establishment ■ demand
4b #7..6	Symmetry ■ bidirectional symmetric		no comparable field
4b #5..1	Information transfer rate (dest.->orig.) ■ - ■ 64 kbit/s		no comparable field

5 #5..1	User information layer 1 protocol ■ no comparable value ■ CCITT V.110 / X.30 ■ CCITT G.711 A-law ■ CCITT X.31 flag stuffing	5 #5..4	Rate adaption ■ no rate adaption ■ V.110 / X.30 rate adaption ■ no comparable value ■ CCITT X.31 flag stuffing
	no comparable field	5 #3..1	Signalling access protocol
	see above	6 #5..2	User information layer 1 protocol ■ default layer 1 protocol
5a #7	Synchronous / asynchronous ■ synchronous ■ asynchronous	6 #1	Synchronous / aynchronous ■ synchronous ■ asynchronous
5a #6	Negotiation ■ not possible	6a #6	Negotiation ■ not possible
5a #5..1	User rate ■ 0,3 kbit/s ■ 1,2 kbit/s ■ 1,2/0,075 kbit/s ■ 2,4 kbit/s ■ 4,8 kbit/s ■ 9,6 kbit/s ■ 12 kbit/s	6a #4..1	User rate ■ 0,3 kbit/s ■ 1,2 kbit/s ■ 1,2 kbit/s / 75 bit/s ■ 2,4 kbit/s ■ 4,8 kbit/s ■ 9,6 kbit/s ■ 12 kbit/s
5b #7..6	Intermediate rate ■ 8 kbit/s ■ 16 kbit/s	6b #7..6	Intermediate rate (6) ■ 8 kbit/s ■ 16 kbit/s
5b #5	NIC on Tx ■ does not require ■ requires	6b #5	NIC on Tx ■ does not require ■ requires
5b #4	NIC on Rx ■ cannot accept ■ can accept	6b #4	NIC on Rx ■ cannot accept ■ can accept
5b #3	Flow control on Tx ■ no		no comparable field
5b #2	Flow control on Rx ■ no		no comparable field
5c #7..6	Number of stop bits ■ 1 bit ■ 2 bits	6a #7	Number of stop bits ■ 1 bit ■ 2 bits
5c #5..4	Number of data bits ■ 7 bits ■ 8 bits	6a #5	Number of data bits ■ 7 bits ■ 8 bits

5c #3..1	Parity information ■ odd ■ even ■ none ■ forced to 0 ■ forced to 1	6b #3..1	Parity information ■ odd ■ even ■ none ■ forced to 0 ■ forced to 1
	no comparable field	6c #7..6	Connection element (1) ■ transparent ■ non-transparent (RLP) ■ both,transp. preferred ■ both,non-transp. preferred
5d #7	Duplex mode ■ half duplex ■ full duplex	4 #4	Duplex mode ■ half duplex ■ full duplex (*)
5d #6..1	Modem type ■ reserved ■ V.21 ■ V.22 ■ V.22bis ■ V.23 ■ V.26ter ■ V.32	6c #5..1	Modem type ■ none (7) ■ V.21 ■ V.22 ■ V.22bis ■ V.23 ■ V.26ter ■ V.32
6 #5..1	User information layer 2 protocol ■ Q.921 (I.441) ■ X.25, link level ■ no comparable value	7 #5..1	User information layer 2 protocol ■ no comparable value ■ X.25, link level ■ IA5 (8)
7 #5..1	User information layer 3 protocol ■ Q.931 (I.451) ■ X.25, packet level		not supported

General notes:

1. Other T/S 46-30 parameter values than those listed in the table, if indicated in the BC-IE, will be rejected by clearing the call.

2. Only the GSM 04.08 parameter values listed in the table may be generated (comparable values) during a mobile-terminated call by mapping the T/S 46-30 parameter values.

3. According to T/S 46-30 and GSM 04.08, respectively, the octets are counted from 1 to n onwards; the bit position in a particular octet is indicated by #x(..y), with {x,y} = 1..8 (1 is the least and 8 the most significant bit).

Notes regarding the mapping:

(*) This GSM 04.08 parameter value is inserted, if the comparable T/S 46-30 parameter value is missing.

(1) This GSM 04.08 parameter value is inserted according to user rate requirements and network capabilities / preferences.

(2) This GSM 04.08 parameter value is inserted, if the information transfer capability in ISDN BC is "3.1kHz audio" and a comparable modem type is specified.

(3) This GSM 04.08 parameter value is inserted, if the information transfer capability is "3.1kHz audio" and the content of the HLC-IE, if any, indicates "facsimile group 2/3". Note that for MAP version 1 the value may be "alternate speech / facsimile group 3 - starting with speech".

(4) The default value according to T/S 46-30 applies.

(5) The T/S 46-30 parameter value "8kHz integrity" will be mapped onto GSM 04.08 parameter value "unstructured".

(6) The GSM 04.08 parameter value is the same as the T/S 46-30 parameter value, if the connection element is "transparent". For any other connection element setting the value is 16 kbit/s, if the radio channel requirements are "full rate" or "both, full rate preferred", or 8 kbit/s, if the radio channel requirements are "half rate" or "both, half rate preferred".

(7) This GSM 04.08 parameter value is inserted, if no modem type is specified or facsimile group 3 is indicated.

(8) Where the network indicates connection elements "non-transparent", "both, transparent preferred" or "both, non-transparent preferred" and the ISDN BC contains no values for the user information layer 2 protocol, then the GSM BC should be forwarded with the user information layer 2 protocol element also containing no value.

10.2.2.1 Circuit switched calls

Where the bearer capability information indicated that the call is a circuit switched unrestricted digital call, then the IWF should select the appropriate rate adapted PLMN bearer service.

10.2.2.2 Packet calls

The mobile network offers only Bm channel access for the packet mode service. The ISDN offers both B and D channel access for the packet mode service. The interworking of mobile packet calls is described in Recommendation GSM 09.06.

10.2.3 Transparent service support (see Rec. 03.10)

Recommendation 08.20 identifies the rate adaptation scheme to be utilised on the BS to MSC link. The transcoding function will re-generate the 64kbit/s rate adapted format utilising the 8 and 16kbit/s intermediate data rates. The MSC - IWF will utilise the same rate adaptation scheme as that indicated in recommendation 08.20, i.e. adapted to 64kbit/s.

10.2.3.1 MSC - IWF rate adaptation scheme

This link consists of a 64kbit/s channel with the information, both user data and in band parameter information (where provided) rate adaptated in conformance to recommendation 08.20.

10.2.3.2 Rate adaptation process in IWF

When interworking to the unrestricted digital bearer service then no further rate adaptation will be necessary within the IWF. When interworking to the 3.1 kHz audio service, then the same process as for the PSTN case is necessary.

10.2.3.3 Mapping of signalling MS/IWF to modem interface requirements

Only necessary for the 3.1 kHz audio restricted interworking case (see section 9.2.3.3).

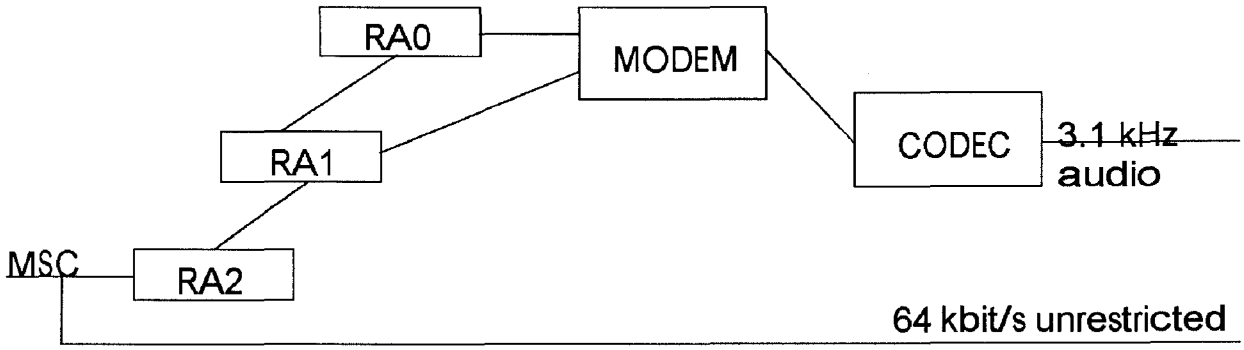


Figure 8
Protocol structure in the IWF (transparent)

10.2.3.4 Establishment of end-to-end terminal synchronizations

Prior to exposing the traffic channel of a PLMN connection to transmission of user data, the controlling entities of the connection have to assure of the availability of the traffic channel. This is done by a so called synchronizations process

- starting on the indication of "physical connection established" resulting from the PLMN-inherent outband signalling procedure

- ending by indicating the successful execution of this process to the controlling entity, which then takes care of the further use of the inband information (data, status).

Network interworking within an IWF is concerned with the terminating side (to the MS) and the transit side (to the fixed network) of a connection. Both sides have to be treated individually related to the synchronizations process.
With respect to the terminating side the procedure is as follows:

- sending of synchronizations pattern 1/OFF (all data bits "1"/all status bits "OFF") to the MS using the RA1/ RA2 rate adaptation function

- searching for detection of the synchronizations pattern 1/OFF from the MS.

When the 1/OFF from the MS has been recognised as a steady state, the IWF continues sending the synchronizations pattern 1/OFF to the MS unless a timer T expires. From this time the information on the receiving lines from the MS and from the fixed network are directly mapped to the respective sending lines.

During the synchronizations process described above, i.e. While the synchronizations pattern is being sent by the IWF, the IWF will not send the V110 frame structure to the ISDN transit network. Once timer "T" expires the synchronizations pattern will continue to be transmitted from the IWF to the MS, however, the IWF will start sending the V110 frames received from the MS to the ISDN transit network. The IWF will start looking for the

ISDN frame alignment to be received from the ISDN. On recognising frame alignment the IWF will cease sending its synchronizations pattern to the MS and connect the ISDN through to the MS.

It should be noted that in a GSM PLMN V.-series and X.-series interfaces are only supported in full duplex mode. Thus the call control phase can be mapped almost completely to the signalling procedure (the S-bits during the call control phase are irrelevant). However, the "ready for data" condition (i.e. CT106/109, in case of V.-series interface, and I-circuit, in case of X.-series interface) is derived directly from the traffic channel (see also filtering of channel control information).

10.2.3.5 Network independent Clocking (NIC)

See also Section 9.2.3.5. The NIC function has to recognise and manage the conversion of the NIC information received incoming from the network. The conversion has to be made to the NIC format used within the GSM System as defined in recommendations 04.21/08.20. The NIC function has to manage the conversion of the GSM NIC format into that used within the ISDN in the traffic direction towards the network.

10.2.4 Non-transparent service support (See rec. 03.10)

Recommendation 08.20 identifies the rate adaptation scheme to be utilised on the BS-MSC link, as for the transparent case.

10.2.4.1 MSC - IWF Rate adaptation scheme

This will be the same as for the transparent case.

10.2.4.2 Protocol layer structure in the IWF

Recommendation 03.10 identifies the protocol layer structure for the non-transparent case, the IWF provides the inverse of the action in the terminal adaptation function.

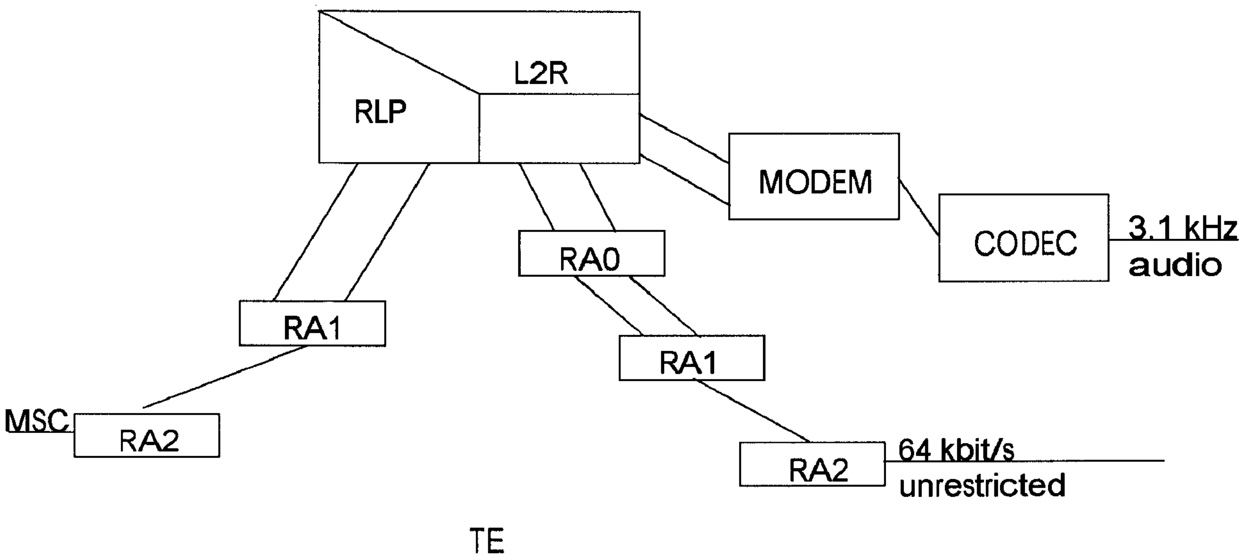


Figure 9
Protocol structure in the IWF (non-transparent)

10.2.4.3 Re-constitution of user data

Recommendation 04.22 refers to the frame of user data in the radio link protocol. The layer 2 relay function in the MS (identified in rec. 03.10) contains the mechanism for the stripping of the start and stop bits, for asynchronous applications, and synchronizations of the user data within the RLP frame. The L2R function within the IWF performs the reverse of this role, for transmitted data by re-inserting the start and stop bits and the stripping of received data.

10.2.4.4 Layer 2 relay functionality

Specific functionality is required on the L2R dependant upon the service which is being requested to be supported. The selection of the appropriate L2R function will be determined by the IWF on the basis of the bearer capability information signalled in the call set-up request, or call confirmation message. The prime information element being transparent or non transparent service indication. In addition the particular L2R function will be selected on the basis of the user layer 2 indicated e.g. X.25, IA5, etc.

The specific interaction between the L2R function and the RLP function and the L2R frame structure will be the same as that detailed in the Annex to the appropriate 07 series recommendation.

10.2.4.5 In band signalling mapping flow control

This entails the L2R function providing the means of controlling and responding to flow control function of the modem (or in the rate adapted frame) plus any synchronizations requirements related to flow control.
Normally, the IWF will utilise flow control by means of the "S" and "X" bits in the V110 frame to the ISDN. Further flow control support will be a network operator option.

Whichever flow control is provided, the L2R function must

(a) provide immediate indication of flow control to the fixed network on receipt of flow control request from the MS.

and/or

(b) provide immediate indication of flow control to the MS on receipt of flow control request from the fixed network i.e. in the next available L2R status octet to be transmitted.

Where in band (X-on/X-off) flow control is in use, then the X-on/X-off characters will not be passed across the radio interface.

10.2.4.5.1 Conditions requiring flow control towards the fixed network

The L2R function will initiate flow control in the following circumstances:

1) The transmit buffer reaches a preset threshold.

2) The L2R function receives a "flow control active" indication.

On removal of buffer congestion receipt of L2R "flow control inactive" the flow control will be removed.

10.2.4.5.2 Conditions requiring flow control towards the MS

The L2R function will transmit to the MS a "flow control active indication" in the following circumstances:

1) If the receive buffer from the radio side reaches a preset threshold.

2) If a flow control indication is received from the fixed network customer. On receipt of this flow control indication, transmission of data from the receive buffers towards the fixed network terminal is halted.

On removal of the buffer congestion or fixed network flow control indication, the L2R function will send a "flow control inactive" indication towards the MS. In addition, for the fixed network indication, transmission of data from the receive buffers will be restarted.

10.2.4.6 Data buffers

10.2.4.6.1 Transmit buffers (towards MS)

Incoming data from the fixed network customer shall be buffered such that if the IWF is unable to transfer data over the radio path the data is not lost.

The buffer shall be capable of holding [16-32 kbits]. When the buffer is half full flow control towards the fixed network shall be initiated as per paragraph 10.2.4.5.1

10.2.4.6.2 Receive buffers (from MS)

Incoming data from the MS is buffered such that if the fixed network terminal is unable to accept the data then it is not lost.

The buffer shall be capable of holding [16-32 kbits] of data. When the buffer becomes half full, the L2R function will send a "flow control active" indication towards the MS, as per paragraph 10.2.4.5.2

10.2.4.7 Transportation of the "Break" condition

The "Break" condition must be recognised by the L2R function and passed immediately to the MS. The L2R function will generate a break condition towards the fixed network on receipt of a break indication from the MS. Information in the buffers will be optionally discarded. (Further study is required regarding the provision of a non-destructive BREAK indication.)

10.2.4.8 Signalling mapping modem status information or "in band" rate adapted frame information

Status information from the modem or within the rate adapted frame, will be carried by the L2R function, in the IWF, to/from the L2R function in the terminal adaptation function. The IWF is not intended to utilise this information for any purpose. The use of "Data carrier detect" or "clear to send" by the terminal adaptation function to determine ISDN link establishment or failure is not utilised by the IWF e.g. call clearing, in event of line failure, will be generated normally by the MS not the IWF.

10.2.4.9 Synchronizations

10.2.4.9.1 V110/ECMA 102 Frame synchronizations

The ISDN frame synchronizations will need to be mapped to the frame synchronizations utilised on the IWF to MSC link. Loss of frame synchronizations either from the ISDN or from the MSC is for further study.

10.2.4.9.2 RLP Frame start indication

The frame start indication is defined in recommendation 08.20. Link establishment and frame error recovery are defined in recommendation 04.22.

10.2.4.9.3 L2R Frame synchronizations

The synchronizations of user data and its interaction between the L2R function and RLP function are defined in Annex to 07 series recommendations.

10.2.4.9.4 Establishment of end-to-end terminal synchronizations

Prior to exposing the traffic channel of a PLMN connection to transmission of user data, the controlling entities of the connection have to assure of the availability of the traffic channel. This is done by a so called synchronizations process

- starting on the indication of "physical connection established" resulting from the PLMN-inherent outband signalling procedure

- ending by indicating the successful execution of this process to the controlling entity, which then takes care of the further use of the inband information (data, status).

Network interworking within an IWF is concerned with the terminating side (to the MS) and the transit side (to the fixed network) of a connection. Both sides have to be treated individually related to the synchronizations process.

With respect to the terminating side the procedure is as follows:

- waiting for RLP link establishment by the MT (in addition the IWF may initiate the RLP link establishment).

The IWF will not send V.110 frame structure to the ISDN transit network and will not start looking for ISDN unless the RLP has been established. On recognising frame alignment the information from/to the RLP is mapped by the L2R entity applicable to this particular bearer capability.

It should be noted that in a GSM/PLMN, V.-series and X.-series interfaces are only supported in full duplex mode. Thus the call control phase can be mapped almost completely to the signalling procedure (the S-bits during the call control phase are irrelevant). However, the "ready for data" condition (i.e. CT106/109, in case of V.-series interface, and I-circuit, in case of X.-series interface) is derived directly from the traffic channel (see also filtering of channel control information).

The synchronizations process described above applies to both the transparent and the non-transparent mode of transmission. However, the exchange and detection of the synchronizations pattern itself is only essential for the transparent mode. Otherwise the RLP automatically assured the availability of the traffic channel.

10.2.5 DTE/Modem interface (Filtering)

The DTEs taken into account for the PLMN at the MS side conform to CCITT's DTE/Modem interface specifications, which assume basically an error-free environment, i.e.

* limited distance, point-to-point local interconnection of the interface circuits for data and status

* steady state signalling.

The envisaged use of these DTE's in the PLMN environment leads to the exposure of these "interconnections" - which may, in the ISDN case, lead to the ISDN Rate Adaptation rather than to a Modem in the IWF - to the PLMN Radio Channel. To assure proper operation even under these conditions appropriate measures have to be taken. In the "non-transparent case" the RLP satisfies the requirement for both data and status lines. In the "transparent" case, the

* data line aspects have to be dealt with end-to-end between the users, while

* status line aspects are of concern to the network which are dealt with in the following.

The use of the channel control information for the remote control of the DTE/Modem control interchange-circuits between the MS and the IWF (the conveyance of which is supported by the rate adaptation scheme adopted for PLMN application) requires alignment to the particular transmission occurrences in the traffic channel to be taken into account within the PLMN. In principle this can be best achieved by

- relying only on the PLMN outband signalling as far as connection control is concerned.

- eliminating the dependence upon the transmission of channel control information via the radio link.

Support for this strategy is given to a certain extent by the confinement of PLMN data connections to

- full duplex operation

- switched service (demand access)

- mapping of connection-control relevant conditions of the DTE/DCE control interchange-circuits to/from outband PLMN signalling according to Rec. GSM 04.08

- flow control supported only in non-transparent mode

- support of connections with the same user data rate only (no TA to TA end-to-end flow control in case of transparent mode).

The only DTE/Modem control interchange-circuit conditions, which actually are not covered by the above confinements, are the indications of readiness for data transmission, i.e. CT106/109 in case of V.-series interface and I-circuit of X.-series interface. As the effect of a condition change of the aforementioned DTE/Modem interchange-circuits depends on the :

- phase within the course of the connection
- direction of change (ON-OFF or OFF-ON)

The required precaution to be applied (Filtering) must be determined individually in view of

- function deduced from the change
- resilience of the connection needed
- error condition possibly invoked due to a delay in performing the condition change of the control interchange circuit
- potential loss of performance in connection usage.

The details of the filtering function are laid down in GSM 07-series Recommendations.

10.3 Interworking Alternate speech data calls

10.3.1 Alternate speech data bearer interworking

10.3.1.1 General

The procedure for the alternate speech unrestricted digital service is implicitly invoked at the call set-up phase. This service is invoked by indication of the "Information transfer capability", alternate speech/unrestricted digital. In addition to this, the call set-up message should include the use of repeated bearer capability messages, one indicating speech and the other indicating the specific data user rate etc., as for normal data calls. The bearer capability first indicated i.e. speech or digital unrestricted determines the first selection required of the network by the subscriber. Optional lower layer and higher layer capabilities may also be included. The IWF should perform both compatibility checking and subscription checking on both sets of capabilities as for normal data calls. If this check fails then the call should be rejected.
The speech capability will utilise the normal telephony teleservice interworking requirements and mobile network capabilities. The unrestricted digital capability will utilise the appropriate data interworking capability and may use either the transparent or non-transparent mobile network capability.

10.3.1.2 Mobile originated ISDN terminated

The call set up is as for the PSTN case. Interworking is provided to the ISDN bearer service 3.1 kHz audio. The modification command will be generated by the mobile subscriber. The message is not transmitted to the ISDN . In this instance it is necessary for change of network capabilities to be carried out in the mobile network.

10.3.1.3 ISDN originated mobile terminated

This handled as for normal ISDN originated call. In the information transfer phase the call is dealt with as indicated in the previous paragraph.

10.3.2 Speech followed by data interworking

10.3.2.1 General

The set up and selection of interworking function for this service is the same as that indicated for the alternate speech digital service. The only difference in this service is that speech will always be the first bearer capability selection and once the modification command is received from the mobile station then all network resources associated with the handling of the speech call may be released for reallocation to other calls, i.e. they will not be required again in the handling of this call. Both mobile originated and terminated are dealt with as detailed in sections 10.3.1.2 and 10.3.1.3.

APPENDIX

Version 1 of the Mobile Application Part (MAP) does not support transfer between the HLR and VLR, and VLR and VMSC of multiple bearer capabilities. In addition, version 1 of the MAP does not support in-call modification following an inter-MSC handover.