

# PT SMG

Permanent

Document

## GSM 10.14

July 1997

Version 1.0.1

---

Source: PT SMG

Reference: PT SMG PD/GSM

**Key words:** Digital cellular telecommunications system, Global System for Mobile communications (GSM)



## **Digital cellular telecommunications system (Phase 2+); System Overview for 14.4 kbit/s Work Item**

**(GSM 10.14 version 1.0.1)**

European Telecommunications Standards Institute

**ETSI PT SMG Secretariat**

**Postal address:** F-06921 Sophia Antipolis CEDEX - FRANCE

**Office address:** 650 Route des Lucioles - Sophia Antipolis - Valbonne - FRANCE

Tel.: +33 492 94 42 66 - Fax: +33 493 65 28 17

Foreword by PT SMG .....	4
1 Objective and Scope of This Paper .....	4
1.1 Introduction .....	4
1.2 Service Performance Requirements for 14.4 kbit/s User Data Services .....	5
2. Performance of Data Services .....	6
2.1 Bit Error Rate vs. Signal to Noise Ratio and Carrier to Interference Ratio .....	6
2.2 Throughput for Non-Transparent Data Services .....	9
2.2.1 Physical Layer Simulations (FER Performance) .....	9
2.2.2 Throughput Simulations .....	11
2.3 Capacity Simulations .....	12
2.3.1 Capacity Simulation Assumptions .....	12
2.3.2 Performance Measure .....	13
2.3.3 Simulation Results .....	13
2.3.4 Transparent Mode .....	14
2.3.5 Non-Transparent Mode .....	14
2.4 Conclusions .....	14
3. Channel Coding .....	15
3.1 General .....	15
3.2 14.5 kbit/s Channel Coding .....	15
3.4 14.4 and 9.6 Interoperability .....	15
3.4.1 Analysis of Handover Delay .....	15
4. Link Protocol .....	16
4.1 Transmission of Status and Control Information .....	16
4.1.1 Multiframe Structure – General .....	16
4.1.2 Multiframe Structure for 14.4 kbit/s Transparent Services .....	16
4.1.2.1 Multiframe Initial Synchronization Procedure .....	17
4.1.2.1.1 Multiframe Resynchronization .....	17
4.1.2.2 Analysis of the Effects of Bit Errors on the Multiframe .....	17
4.1.2.2.1 Multiframe Synchronization Pattern Bits (M1 Bitstream) .....	17
4.1.2.2.2 Status/Control Bits (M2 Bitstream) .....	17
4.1.2.2.2.1 HSCSD Subchannel Numbering Bits .....	17
4.1.2.2.2.2 Terminal Status Bits .....	17
4.1.2.2.2.3 Network Independent Clocking Bits .....	17
4.1.3 Multiframe Structure for Non-Transparent Services .....	18
4.2 Non-Transparent Service .....	18
4.2.1 Different RLP Version Types .....	18
4.2.1.1 Delay Analysis for 14.4 kbit/s RLP Frame Structure: .....	19
4.2.2 Resetting the RLP Structure – “REMAP” Procedure .....	19
4.2.3 Modifications to L2R Addressing .....	20
4.2.4 Analysis of Asynchronous Non-Transparent Data Rates .....	21
4.2.5 Analysis of Synchronous Non-transparent Data Rates .....	23
4.3 Transparent Service .....	25
4.3.1 Synchronous Transparent Data .....	25
4.3.1.2 Analysis of Synchronous Transparent Data Rates .....	26
4.3.2 Asynchronous Transparent Data .....	28
4.3.2.2 Analysis of Asynchronous Transparent Data Rates .....	29
4.3.3 Facsimile Service .....	31
4.3.3.1 Analysis of Facsimile Data Rates .....	32
Annex A: Structure of the Extended-TRAU (E-TRAU) Frame Format used for 14.4 kbit/s Data Transmission on the <i>Abis</i> -Interface .....	34
A.1 Synchronization of the <i>Abis</i> -Interface for 14.4 kbit/s Services .....	34

A.2	14.4 kbit/s Services Data Transmission on the <i>Abis</i> -Interface.....	36
Annex B:	Structure of the A-TRAU Frame Format used for 14.4 kbit/s Data Transmission on the A-Interface .....	38
B.1	Zero Sequence Position (ZSP) Procedure.....	39
B.1.1	ZSP Procedure Description.....	39
B.1.2	ZSP Procedure Example .....	40
Annex C:	Approved Change Requests pertaining to 14.4 kbit/s Workitem.....	43

## Foreword by PT SMG

The main part of this document was originally developed by T1P1.5 as part of the joint SMG - T1P1.5 "14.4 kbit/s" Work item, where T1P1 had the prime responsibility. It should be noted that due to the history of this document the simulation results presented pertain to the GSM1900 and are not necessarily the same for other frequency bands. Other specific references to GSM1900 should be read as referring to GSM in general.

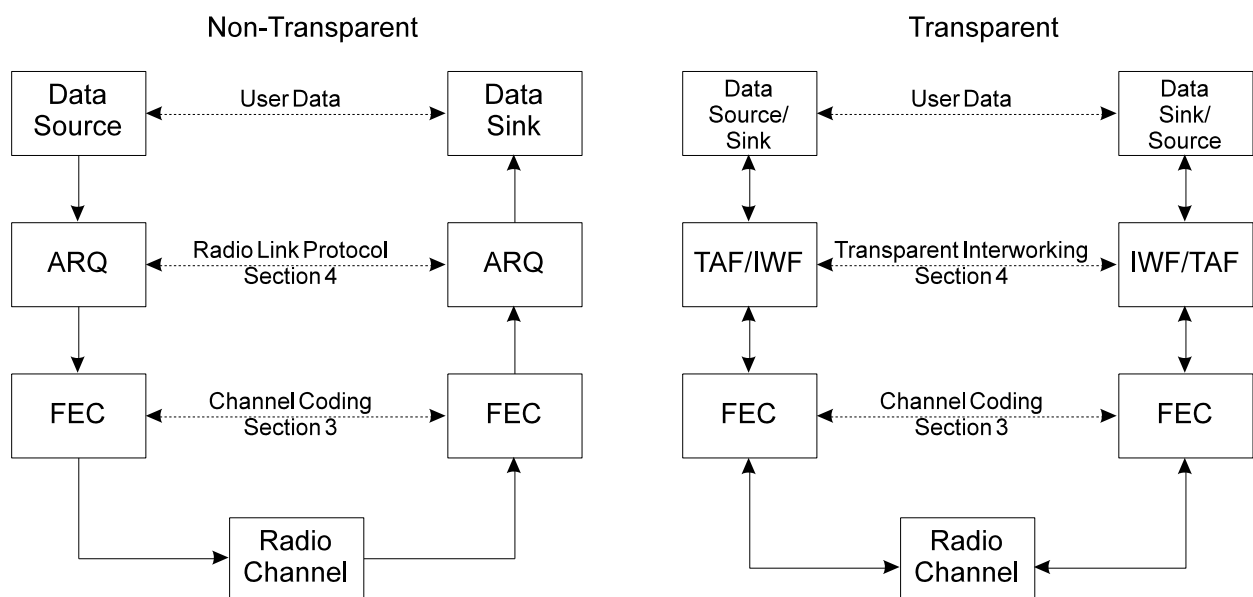
Further, a list of change requests that are approved by TC SMG pertaining to the workitem is maintained in Annex C.

## 1 Objective and Scope of This Paper

This paper presents a system overview of the 14.4 kbit/s data services in GSM. Solutions for both non-transparent and transparent services are included.

To clarify, transparent service is a constant speed, variable error rate service using only forward error correction (FEC) between the MS and BTS to maintain data integrity. Non-transparent service is a variable bit rate (variable throughput) service which uses FEC between the MS and BTS, and also makes use of a Radio Link Protocol (RLP) between the TAF and IWF to detect and provide retransmission's of damaged or lost frames.

14.4 kbit/s service changes three areas of the "stack" as depicted in Figure 1. Changes to the lowest layer, channel coding, are described in Section 3. Changes to the radio link protocol, ARQ, are described in Section 4.



**Figure 1 - Stack Model for GSM Data Services.**

### 1.1 Introduction

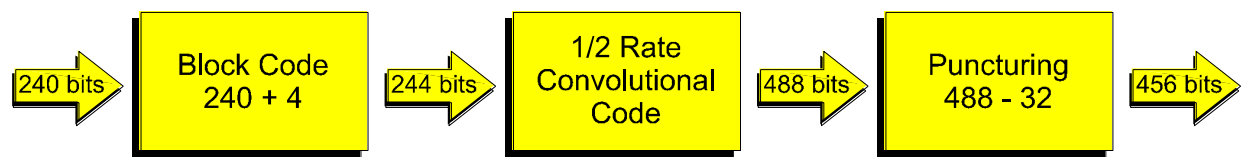
Definitions:

- "Radio Interface User Rate" is defined as the transfer rate for user data (i.e., data usable by the application) over the radio interface.
- "Radio Interface Rate" is defined as the radio interface user rate plus additional overhead for the transmission protocols.

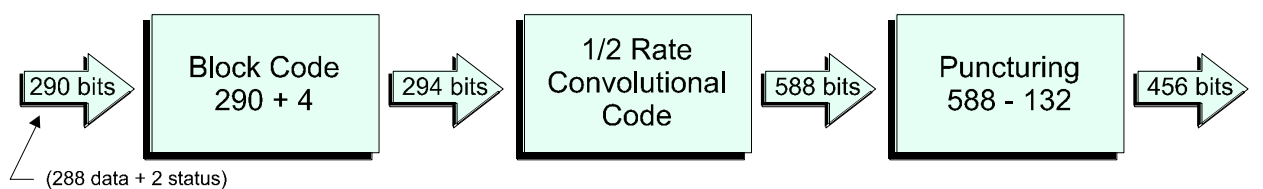
This paper presents a system overview of single slot 14.4 kbit/s data services in GSM. The motivation for this service is the need to maintain a technological advantage over competing wireless technologies in the U.S. by increasing user bandwidth with minimum affect on the operator's capacity. The approach has been optimised for use with V.32*bis* and V.34 wireline modems including V.42 error correction, all commonly used data modem standards.

At present, GSM supports a 12 kbit/s radio interface rate with a 9.6 kbit/s radio interface user rate. By changing the puncturing scheme (see figure 2), a 14.4 kbit/s radio interface user rate is achieved with minimum changes to the GSM system.

**12 kbps (Current GSM1900 Service)**



**14.5 kbps (14.4 kbps Radio I/F User Rate)**



**Figure 2 - Channel coding**

**1.2 Service Performance Requirements for 14.4 kbit/s User Data Services**

This section defines the service and performance requirements for 14.4 kbit/s services. Requirements are levied in the following areas:

- Signaling Channels
- Service Deployment and Handovers
- Transparent Data Service
- Non-Transparent Data Service
- 14.4 kbit/s Modem Service
- 14.4 kbit/s ISDN Service
- 14.4 kbit/s Facsimile Service
- *Abis*-interface Channel Rate
- *A*-Interface Channel Rate

**Signaling Channels**

- Implementation of 14.4 data services shall not affect layer 1 and layer 2 of the signaling channels.

**Service Deployment and Handovers**

- Selective-cell-deployment of 14.4 kbit/s data services within a network shall be supported. This feature allows the operator to partially deploy 14.4 data services in those cells where it can be justified on a cost/benefit basis, and avoids the need for full deployment.
- To support selective-cell-deployment, handover to a cell supporting a maximum of 9.6 kbit/s shall only be required for non-transparent data services.

**Transparent Data Service**

- A 14.4 kbit/s radio interface user rate shall be supported for all transparent data services.

### Non-Transparent Data Service

- A 13.0-13.2 kbit/s radio interface user rate shall be supported for all non-transparent data services.

### 14.4 kbit/s Modem Service

- 14.4 kbit/s modem service shall support transparent and non-transparent modes of operation, using both asynchronous and synchronous data services.
- 14.4 kbit/s modem service shall be compatible with V.32*bis* and V.34 wireline modems.

### 14.4 kbit/s ISDN Service

- 14.4 kbit/s ISDN service using V.110 shall support transparent and non-transparent modes of operation, using both asynchronous and synchronous data services.
- 14.4 kbit/s ISDN service using V.120 shall only be required to support the non-transparent mode of operation, using both asynchronous and synchronous data services.

### 14.4 kbit/s Facsimile Services

The only required mode of operation for 14.4 kbit/s facsimile services shall be synchronous transparent.

- 14.4 kbit/s facsimile data rate shall be supported.

### Abis-interface Channel Rate

- All 14.4 kbit/s data and facsimile services shall fit within a 16 kbit/s channel on the *Abis*-interface.

### A-Interface Channel Rate

- All 14.4 kbit/s data and facsimile services shall fit within a 16 kbit/s channel on the A-interface.

## 2. Performance of Data Services

### 2.1 Bit Error Rate vs. Signal to Noise Ratio and Carrier to Interference Ratio

Figures 3a to 3b compare the simulated performance of channel coding schemes for 12 and 14.5 kbit/s radio interface rates (9.6 kbit/s and 14.4 kbit/s radio interface user rates, respectively). The graphs show the coded bit error rate (BER) performance as a function of the signal to noise ratio (SNR) and of the carrier to interference ratio (C/I). The simulation has been done with a TUx (typical urban mobile, at x km/h) channel as defined in GSM 05.05. Modelling includes: no frequency hopping or ideal frequency hopping, mobile speed of 3 km/h or 50 km/h, all at a frequency of 1900 MHz. Only graphs for the best case (TU50 - ideal frequency hopping) and for the worst case (TU3 - no frequency hopping) are included. The graphs for the other cases (TU3 - ideal frequency hopping; TU50 - no frequency hopping) show essentially the same results as TU50 - ideal frequency hopping.

As indicated in the figures 3a and 3b, the puncturing scheme for the 14.4 kbit/s (middle) curve provides a performance level which is about 2 dB less than that of 9.6 kbit/s (leftmost) curve. This reduction in performance was deemed acceptable by the operators of GSM North America.

In the case where there is no frequency hopping and the vehicle speed is 3 km/h (figure 3b), we have the worst case scenario and the performance for all data is relatively low. In the case where there is frequency hopping and the vehicle speed is 50 km/h (figure 3a), we have the best case scenario and the performance for all data rates is substantially better. By modelling both we effectively "bracket" the best and worst case possibilities. Actual network implementation will result in performance somewhere in-between, and is dependent upon network design.

**Page 7**

**GSM 10.14 version 1.0.1: July 1997**

Nokia studies presented at the Personal Indoor Mobile Radio Communication (PIMRC) '96 Conference, comparing the coverage of 14.4 kbit/s service against 9.6 kbit/s (TCH/F9.6), indicate that if the current [TCH/F9.6] service has 90% coverage, then the proposed 14.4 kbit/s service will have about 84-87% coverage. This coverage level was deemed acceptable by the operators of GSM North America.

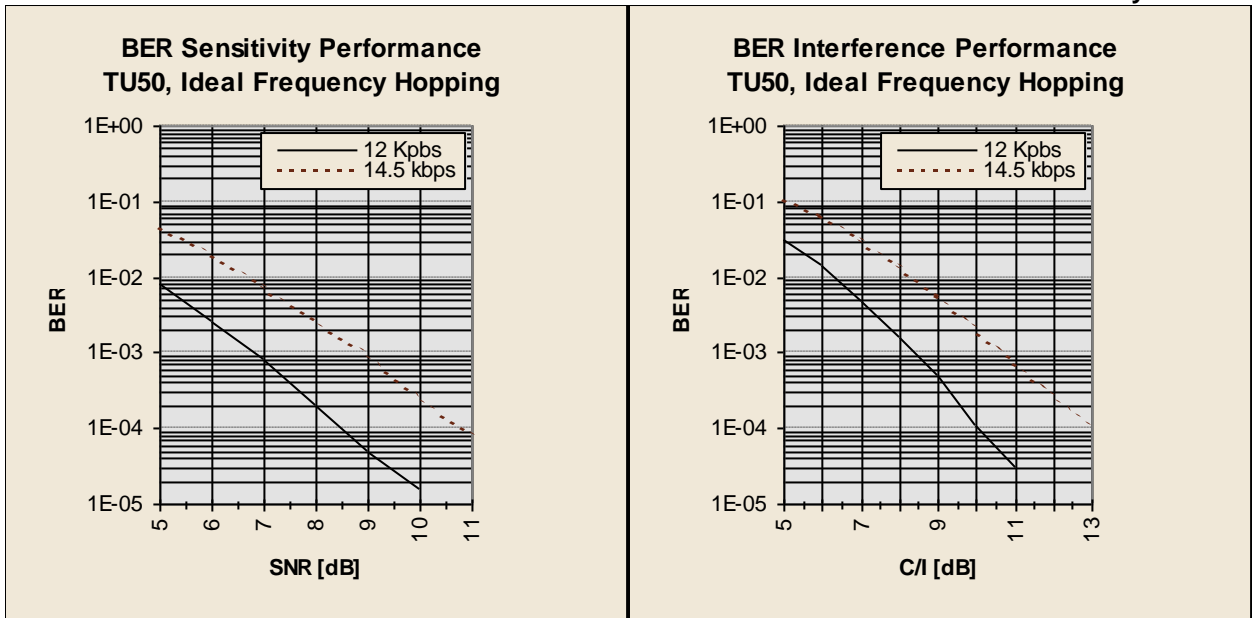


Figure 3a - Comparison of channel coding performance of the 12 and 14.5 kbit/s radio interface rate schemes with Ideal Frequency Hopping and at 50 km/h vehicle speed.

TU 3 - ideal frequency hopping, and TU50 - no frequency hopping (graphs not shown) have been shown to provide performance curves very similar to figure 3a.

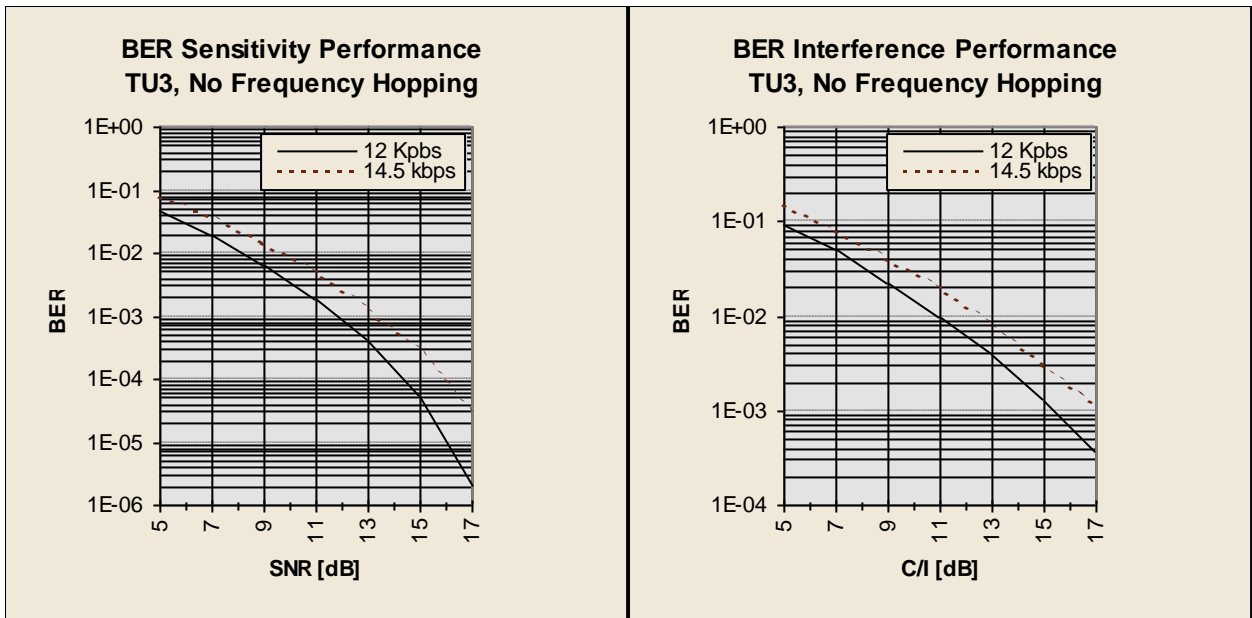


Figure 3b - Comparison of channel coding performance of the 12 and 14.5 kbit/s radio interface rate schemes with No Frequency Hopping and at 3 km/h



## 2.2 Throughput for Non-Transparent Data Services

As described later in Section 4.2, a new "double-RLP" frame structure has been created for 14.4 kbit/s non-transparent data services. This new structure extends the frame length from 240 bits to 576 bits. The increase in frame length, along with the increased BER for 14.4 service shown in the preceding section, has an impact on the number of retransmission's for 14.4 non-transparent service. The number of retransmission's is dependent upon the  $C/I$  for the 14.4 service. The analysis in this section was performed to quantify the effects of increased BER and increased frame length on the overall throughput for non-transparent services.

### 2.2.1 Physical Layer Simulations (FER Performance)

An initial output from the physical layer simulations is the BER. For NT-data, the framer error rate (FER) is important also. The FER was calculated by taking  $n$  consecutive bits out of the BER evaluation block, where  $n$  corresponds to the frame length (240 bits for 9.6 service, 576 bits for 14.4 service). A frame error occurs if there is an error in one or more bits within the frame.

Figures 4a and 4b show the effects of noise and interference on FER depending on speed and whether frequency hopping is used.

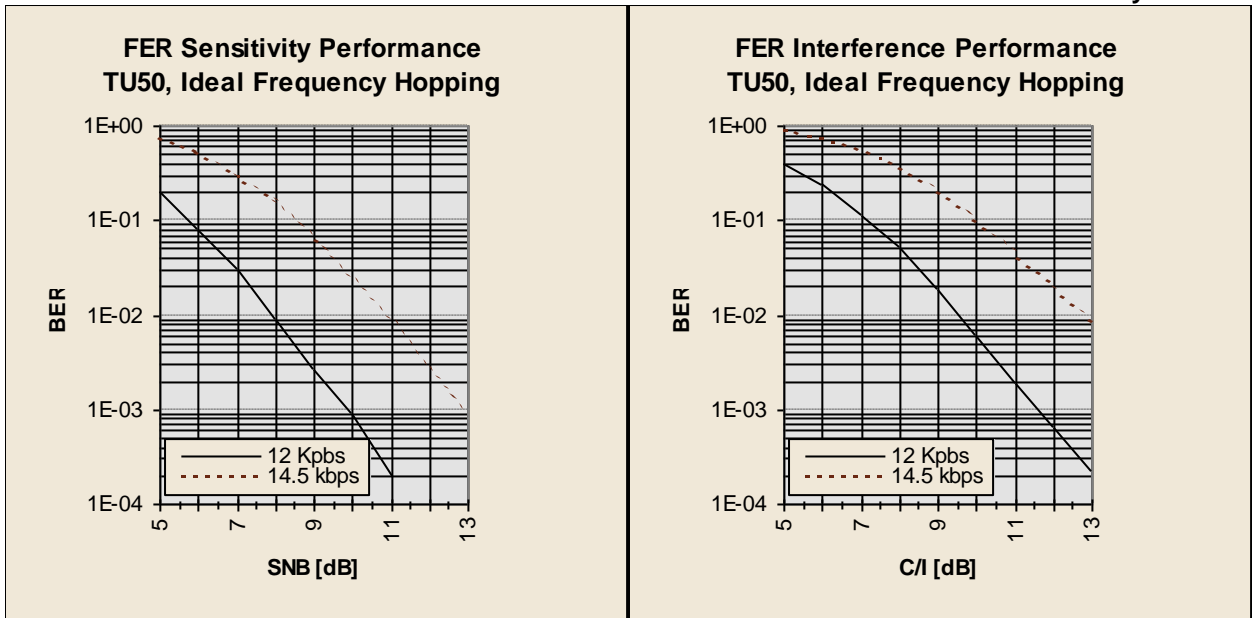


Figure 4a - Comparison of channel coding performance of the 12 and 14.5 kbit/s radio interface rate schemes with Ideal Frequency Hopping and at 50 km/h vehicle speed.

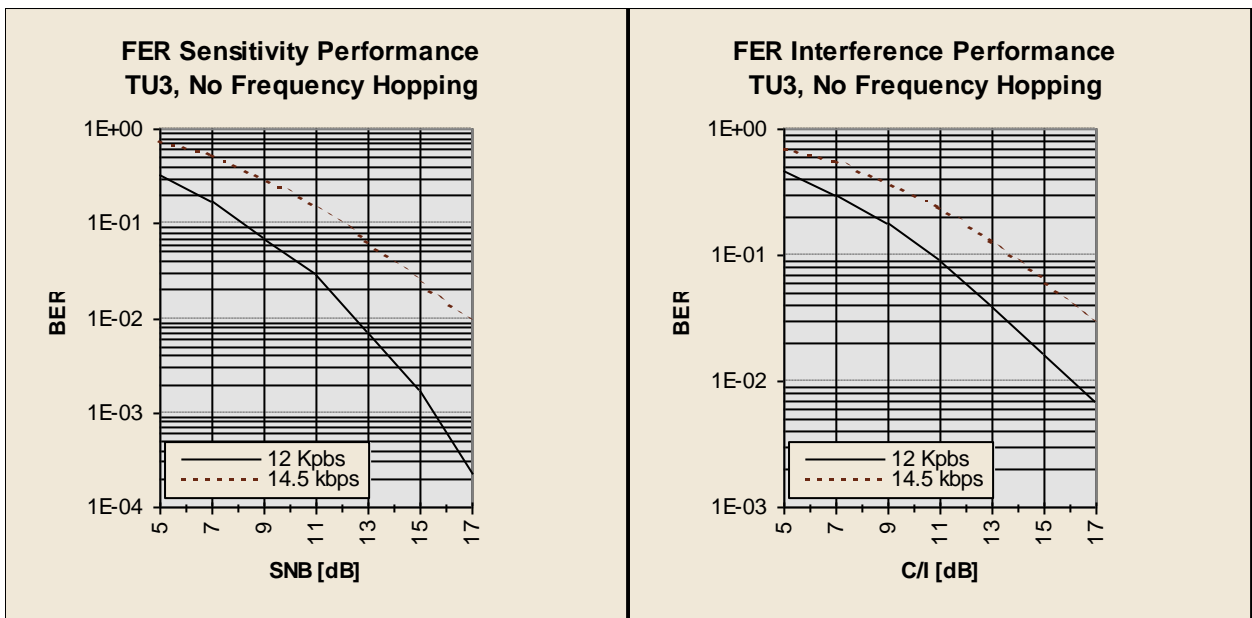


Figure 4b - Comparison of channel coding performance of the 12 and 14.5 kbit/s radio interface rate schemes without Frequency Hopping and at 3 km/h vehicle speed.

Based upon the computed FER's, it is possible to compute the throughputs associated with non-transparent services at various signal to noise and C/I ratios. These graphs are shown in figures 5a and 5b. As these graphs, show, much higher throughputs can be obtained using the 14.5 channel coding.

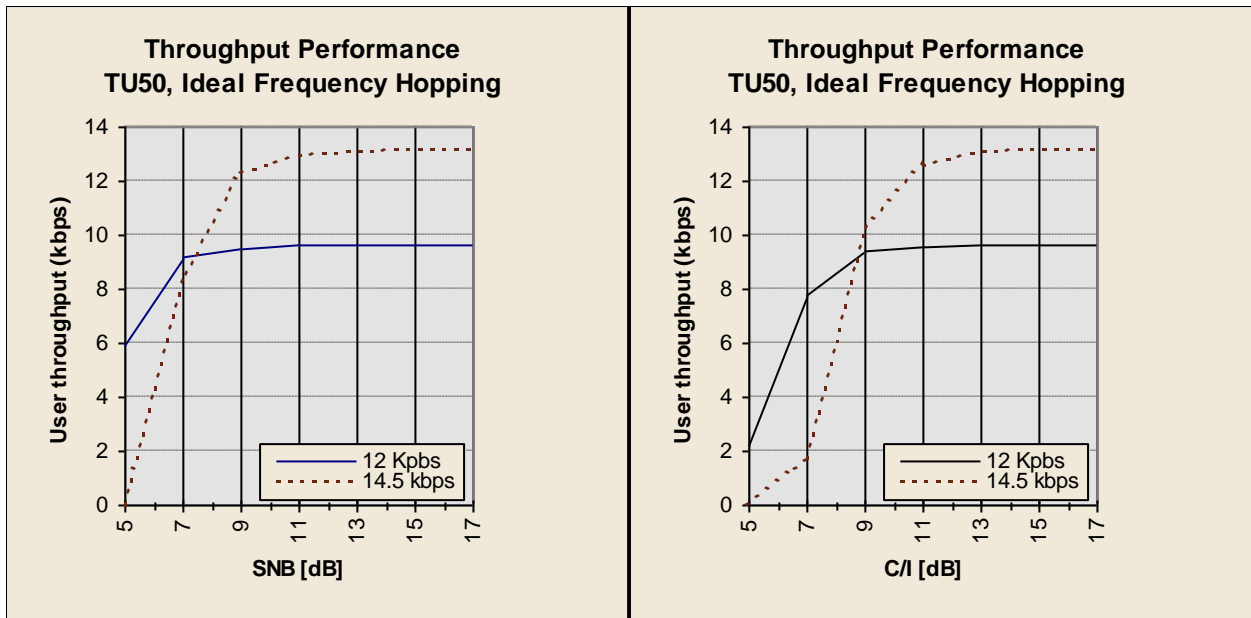


Figure 5a - Comparison of throughput performance of the 12 and 14.5 kbit/s radio interface rate schemes with Ideal Frequency Hopping and at 50 km/h vehicle speed.

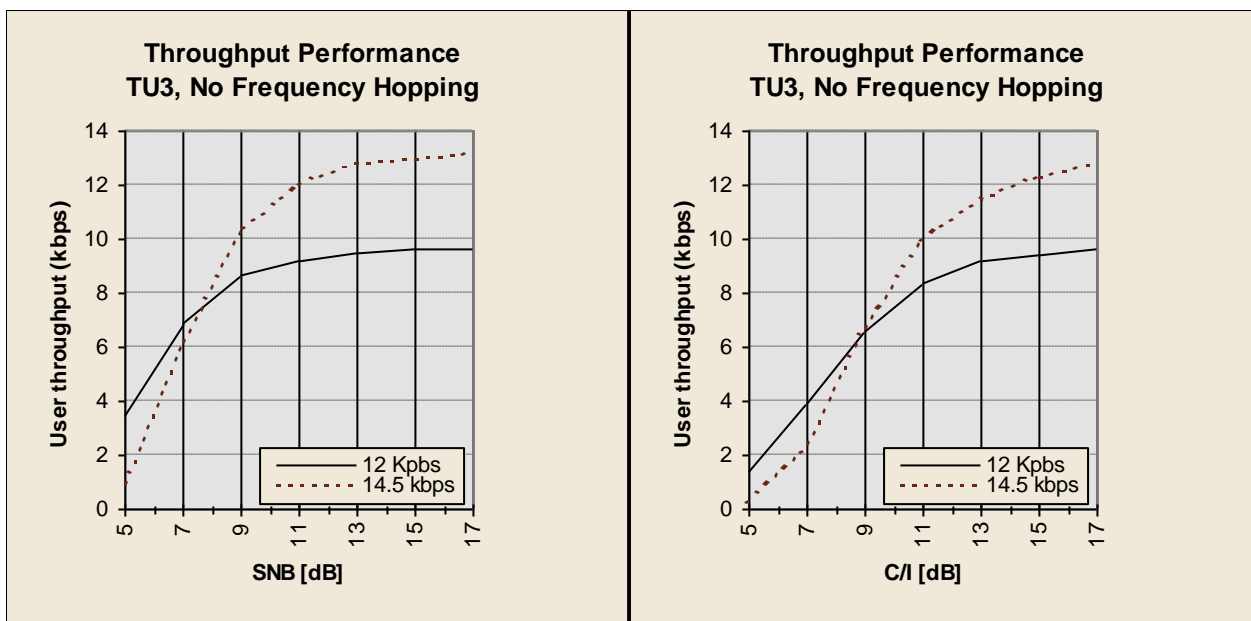


Figure 5b - Comparison of throughput performance of the 12 and 14.5 kbit/s radio interface rate schemes without Frequency Hopping and at 3 km/h vehicle speed.

## 2.3 Capacity Simulations

The previous sections analyzed how error rates and throughput varied depending on SNR and C/I. In order to predict the viability of the 14.5 channel coding, it is necessary to understand what the distribution of C/I values might be in a typical network. It is useful to do this over a simulated network and not a single cell as the results will vary due to cell overlap.

In this section a simple capacity study for the existing 12 kbit/s and the 14.5 kbit/s coding schemes is presented. Networks vary widely, and this is only a simulation. These results should therefore only be used to compare the 12 kbit/s and the 14.5 kbit/s channel coding schemes and not as an estimate of true capacity. Note that the results hold only under the assumptions described below.

The coding schemes are studied in a 3/9-reuse system, using frequency hopping at a served load of 82%. This corresponds to a blocking probability of 2%. Further, a bandwidth of roughly 15 MHz, allocated to the downlink, is assumed. Only the downlink connections are considered in this study. Finally, no power control is utilized.

### 2.3.1 Capacity Simulation Assumptions

Most of the assumptions used in the system simulations are presented in Table 1. It is also worthwhile to point out the following:

- The cell size was chosen small enough to make the thermal noise negligible.
- There was no adjacent channel interference assumed.

**Table 1 System simulation parameter assumptions**

Simulation type	Static (uncorrelated snapshots)
Cell plan	16 clusters à 9 cells, 3 sites per cluster with 3-sector antennas.
Number of base stations, B	144
Distance attenuation	$35\log(d)$
Lognormal shadow fading standard deviation	6 dB
Mobile geographical distribution	Uniform within system
Offered Traffic load	Offered load such that a served load of 82% is achieved. This corresponds to a blocking probability of 2%
Cell selection	Based on least path loss
Handover hysteresis	3 dB
Total bandwidth (downlink), W	14.4 MHz
Number of Traffic channels per cell	64
Channel allocation to cells	Fixed 3/9 reuse pattern
Channel allocation within cells	Random
Power control	No power control
Frequency hopping sequences	Uncorrelated between cells, orthogonal within cells.

The output of the system simulation performed was a large set of C/I ratios, one ratio per user and snapshot. The set of C/I values can be mapped into two sets of bit error rates, one for each coding scheme, and two sets of frame error rates. The link performance results from the TU50-FH case were used. For C/I ratios larger than the range where results of the link performance are available, the corresponding BER and FER have been set to zero. However, this will not affect the results significantly.

### 2.3.2 Performance Measure

The performance measure for both the transparent and the non-transparent mode is spectrum efficiency,  $\nu$ , in terms of kbit/s/cell/MHz.

- In the transparent mode,  $\nu$  obtained as:

$$(1) \nu = \frac{R_{ai} \cdot \omega}{BW} \text{ [kbit/s/cell/MHz]}, \text{ where}$$

$R_{ai}$  is the radio interface bit rate (12/14.5 kbit/s)  
 $\omega$  is the total number of users of all snapshots that has a BER <  $\gamma_{BER}$ ,  
 $\gamma_{BER}$  is a specific BER threshold (varied in figure 7).  
 $B$  is the number of base stations,  
 $W$  is the total bandwidth.

- In the non-transparent mode,  $\nu$  is obtained as:

$$(2) \nu = \frac{\sum_{i=1}^N R_u (1 - FER_i)}{BW} \text{ [kbit/s/cell/MHz]},$$

where

$N$  is the total number of user of all snapshots,  
 $R_u$  is the user bit rate (9.6/13.2 kbit/s),  
 $FER_i$  is the frame error rate for user  $i$ .

### 2.3.3 Simulation Results

In Figure 6, both uplink and downlink C/I distributions are shown. Even though the uplink is of no interest here, it should be noted that the uplink performance is about 4 dB worse at the 10% level. However, this will be compensated for by using receiver antenna diversity in the uplink.

#### C/I Distribution

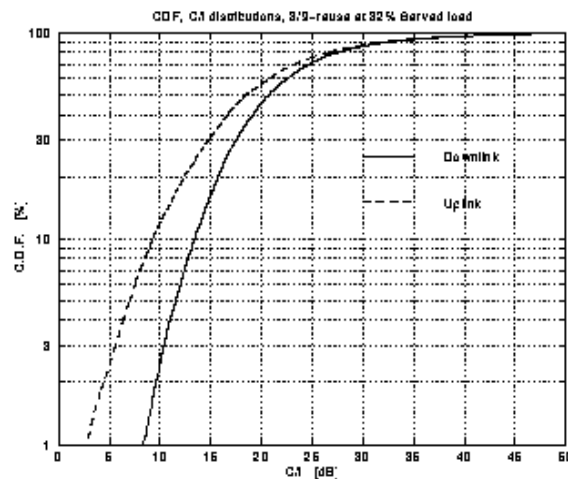


Figure 6 - CDF's of the uplink and downlink C/I distributions

Figure 6 shows that most of the C/I ratios are above the critical BER and FER intervals (see Figures 3-5). These high C/I ratios imply that the system is blocking limited, rather than interference limited, (i.e., the capacity depends on the number of channels). Note that for other reuse patterns, the results would

be different. Note also that no power control is included. Power control would make it possible to merge the C/I ratios around a certain value and could thus be a benefit to the 14.5 kbit/s coding scheme.

### 2.3.4 Transparent Mode

In Figure 7a the spectrum efficiency for different BER requirements is shown. The figure shows that the 14.5 kbit/s coding scheme has the better spectrum efficiency for all the BER requirements shown. This is explained by the high C/I ratios, which result in low BER for both coding schemes. The average bit error rates were:

$$BER_{12} = 1.71 \cdot 10^{-4} \qquad BER_{14.5} = 6.28 \cdot 10^{-4}$$

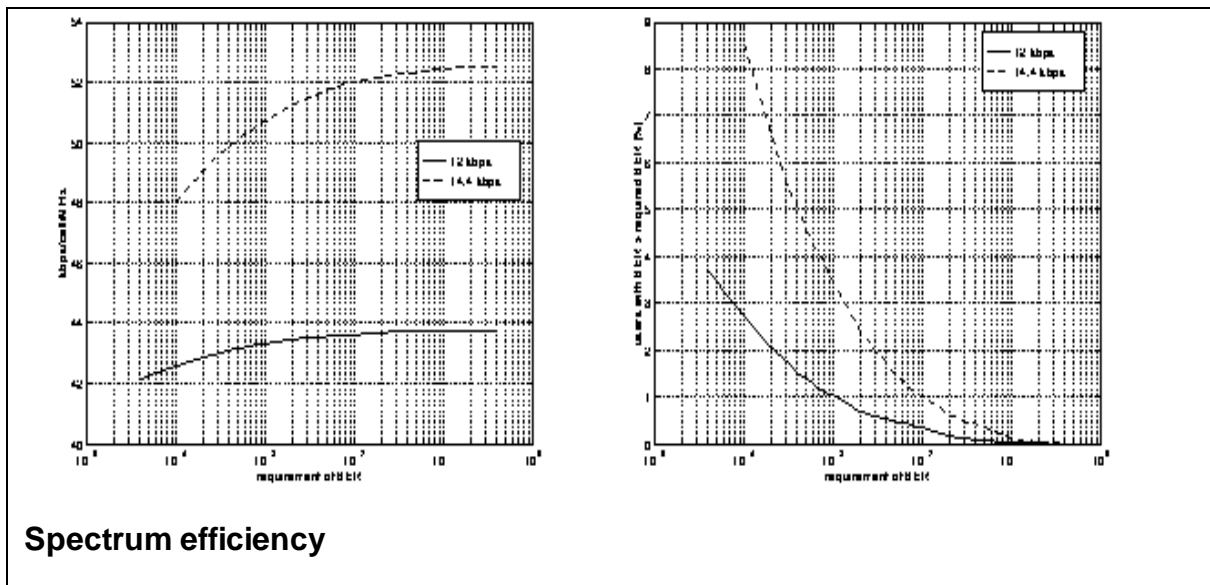


Figure 7a – Spectrum efficiency for the two coding schemes for different BER requirements.  
 Figure 7b – Percentage of the users that have a larger BER than required BER

It should be noted that the fraction of users who did not satisfy the BER constraint do not contribute to the spectrum efficiency shown. These users do however get bits through the system and the results are thus pessimistic. How large this fraction of users is depends on the BER requirement. This dependence is shown in Figure 7b.

### 2.3.5 Non-Transparent Mode

In this mode only correctly received frames contribute to the spectrum efficiency. Table 2 shows the results for the non-transparent mode. The spectrum efficiency and the average FER are shown.

Table 2 Results, non-transparent mode

Coding scheme	User bit rate, $R_u$	Average FER	Spectrum efficiency [kbit/s/cell/MHz]
12 kbit/s	9.6 kbit/s	0.0022	34.94
14.5 kbit/s	13.2 kbit/s	0.0108	47.63

The results show that although the average FER is larger, the 14.5 kbit/s coding scheme results in larger spectrum efficiency.

## 2.4 Conclusions

The simulation results show that the 14.5 channel coding provides adequate data robustness and improved throughput in most environments. The 14.5 kbit/s channel coding provides greater spectral efficiency than the existing 12 kbit/s channel coding for both transparent and non-transparent dates services.

### 3. Channel Coding

#### 3.1 General

When designing the new 14.5 kbit/s channel coding, emphasis was placed on minimizing changes to the network infrastructure.

#### 3.2 14.5 kbit/s Channel Coding

Blocks of 290 information bits are delivered to the channel codec every 20msec,  $\{d(0), \dots, d(289)\}$ , and four tail bits are added:

$$\begin{aligned}u(k) &= d(k) & k &= 0, \dots, 289 \\u(k) &= 0 & k &= 290, \dots, 293 \text{ (tail bits)}\end{aligned}$$

The resultant block of 294 bits  $\{u(0), \dots, u(293)\}$  is encoded with a  $\frac{1}{2}$  rate convolutional code defined with the following polynomials (identical to TCH/F9.6):

$$G_0 = 1 + D^3 + D^4$$

$$G_1 = 1 + D + D^3 + D^4$$

Resulting in 588 coded bits  $\{C(0), C(1), \dots, C(587)\}$ .

The code is punctured so that 132 bits are not transmitted. The result is a block of 456 coded bits,  $\{c(0), \dots, c(455)\}$ . The puncturing scheme can be described in terms of the following equation for  $C(n)$ , where the  $C(n)$  bits are not transmitted:

$$C(n) = C(18*j + 1) + C(18*j + 6) + C(18*j + 11) + C(18*j + 15); \text{ for } j = 0, 1, \dots, 31$$

In addition,  $C(577)$ ,  $C(582)$ ,  $C(584)$ , and  $C(587)$  are also punctured.

Interleaving remains as defined in GSM 05.03 for TCH/F9.6 service.

Please refer to GSM 05.03 for more information on the channel coding.

#### 3.4 14.4 and 9.6 Interoperability

Handover interoperability between 14.4 and 9.6 services will only be required for non-transparent services. In both handover cases ( $14.4 \rightarrow 9.6$ , and  $9.6 \rightarrow 14.4$ ), the handover command message will be used for indicating the new channel mode to the mobile station.

As explained further in section 4.2, the RLP frame structures are different for 14.4 and 9.6 services. When moving between cells that support 14.4 and cells that only support 9.6, it will be necessary to reset the RLP used with the previous cell, and begin using the new RLP. Note that any data that was transmitted with the previous RLP, which has not been correctly received, must be retransmitted with the RLP negotiated for the new cell. The procedure for resetting the RLP structure is referred to as a "REMAP", and is detailed in section 4.2.2.

##### 3.4.1 Analysis of Handover Delay

For handover from a 14.4 cell to a 9.6 cell, the interleaving delay before resuming transmission is the same as for a  $9.6 \rightarrow 9.6$  handover ( $\approx 110$ msec).

For handover from a 9.6 cell to a 14.4 cell, there is an additional 20msec of interleaving delay due to the double-RLP frame length (total interleaving delay  $\approx 130$ msec).

For handovers that involve changing the RLP frame length (e.g., 14.4 ↔ 9.6), an additional delay will be incurred to perform the REMAP procedure described in section 4.2.2. This delay is ≈400msec for the mobile end, and ≈600msec for the network end.

## 4. Link Protocol

### 4.1 Transmission of Status and Control Information

#### 4.1.1 Multiframe Structure – General

The basic data services in GSM (e.g., 9.6 kbit/s, 4.8 kbit/s) use the data traffic channel to transfer both user information, as well as status and control information (e.g., terminal status bits; Network Independent Clocking (NIC) bits). For 14.4 kbit/s data services, a multiframe structure is instead used to transfer the status and control information. This approach reduces the overhead in the channel and allows for true 14.4 kbit/s transmission.

In the multiframe structure, status and control information is time-multiplexed over several successive frames. Two bits are used over the radio interface, with each 288-bit data block, to transport the multiframe information. In the network, two bits in the E-TRAU and A-TRAU frame structures (see Appendices A&B) are used to transport the information between the BTS and IWF. The combination yields an end-to-end multiframe structure between the TAF and IWF. The multiframe structure is different for transparent and non-transparent services, as explained in sections 4.1.2 and 4.1.3.

#### 4.1.2 Multiframe Structure for 14.4 kbit/s Transparent Services

The multiframe for 14.4 kbit/s transparent service is structured as follows:

- The first bit (M1) is used to form a 31-bit PN multiframe synchronization pattern (0000 1001 0110 0111 1100 0110 1110 101).
- The second bit (M2) is used to transport the multiplexed status and control information shown below:
  - Subchannel numbering (#)
  - V.24 Terminal Status (SB, X)
  - Network Independent Clocking Code (N)

The multiframe structure is shown in Figure 8 below:

bit number	0 1 2 3	4 5 6 7	8 - 11	12- 15	16 - 19	20 - 23	24 - 27	28 - 30
M1:	0 0 0 0	1 0 0 1	0 1 1 0	0 1 1 1	1 1 0 0	0 1 1 0	1 1 1 0	1 0 1
M2:	# # # SB	SB X # #	# X SB SB	# # # SB	SB X # #	# X SB SB	NNNN	N SB SB

**Figure 8 – Multiframe Structure for 14.4 kbit/s Service**

As can be seen in the figure, the subchannel numbering and terminal status information are repeated multiple times per multiframe. This has been done to reduce the time needed to regain the information carried by the M2 bitstream if multiframe synchronization is lost (e.g., after a handover).

The ### bits carries the subchannel numbering information needed for the multiple substreams used in HSCSD operation.

The SB bits carry V.110 SB-status information (CT 109 for V-series interface, and I-circuit for X-series interface). The X bits carry V.110 X-status information (CT 106 for V-series interface, and N/A for X-series interface).

Note: SA terminal status is not carried in the multiframe. SA status information is not derived directly from the traffic channel, it is generated locally at the TAF/IWF.



The five N bits carry the network independent clocking code that is used to make periodic adjustments to the system clocking. The transmission of the NIC code once per multiframe is sufficient to meet the 100 PPM (part per million) timing requirements in the GSM system. This is shown below:

- Timing requirement of 100 PPM = a minimum compensation of one bit for every 10000 bits that are transmitted ( $100/1000000 = 1/10000$ ).
- Total number of bits transmitted during one multiframe = 320 bits/subframe \* 31 subframes = 9920 bits. Therefore, one NIC compensation can be carried out during every multiframe and meet the 100 PPM timing requirements.

#### **4.1.2.1 Multiframe Initial Synchronization Procedure**

Synchronization to the multiframe is based on the unique PN properties of the 31-bit M1 synchronization pattern (i.e., any 5 bits in the M1 bitstream form a unique pattern that is sufficient to locate the current position within the multiframe).

During the initial synchronization phase, the M1 bitstream is searched for a unique 5-bit pattern needed to locate position within the multiframe. During this time the M2 terminal status information bits (SB, X) are in the OFF condition, and the ### bits carry the subchannel information.

Once the position within the multiframe has been established, the terminal status information may be received from the M2 bitstream.

##### **4.1.2.1.1 Multiframe Resynchronization**

In general, once the multiframe has been synchronized, synchronization should not be lost unless the GSM radio interface synchronization is also lost. If the radio interface synch is lost (e.g., due to a handover), the multiframe synchronization procedure will begin again as soon as the radio frame synchronization is regained. As noted above, this procedure happens quickly (5 bit-times = 100msec).

#### **4.1.2.2 Analysis of the Effects of Bit Errors on the Multiframe**

##### **4.1.2.2.1 Multiframe Synchronization Pattern Bits (M1 Bitstream)**

The multiframe pattern was chosen to be a PN sequence so that only 5 bits would be needed to uniquely identify the position within the overall multiframe. In the event of a bit error(s) in the multiframe pattern, synchronization will be regained in no more than 100msec.

##### **4.1.2.2.2 Status/Control Bits (M2 Bitstream)**

###### **4.1.2.2.2.1 HSCSD Subchannel Numbering Bits**

The condition of the subchannel numbering remains constant, and the bits themselves are repeated continuously within the multiframe. The combination makes the reception of the subchannel numbering inherently tolerant of bit errors.

###### **4.1.2.2.2.2 Terminal Status Bits**

This first level of protection against bit errors is provided by continuously repeating the terminal status bits (SB, X) throughout the multiframe as shown earlier. A second level of protection is provided by requiring an integration period for the bit values – before a change of state is made based on the value of the SB or X bits, at least two consecutive SB or X bits, respectively, must carry the same value.

###### **4.1.2.2.2.3 Network Independent Clocking Bits**

The NIC code is protected against bit errors as described in TS 04.21.

### 4.1.3 Multiframe Structure for Non-Transparent Services

For non-transparent services, the status and subchannel numbering information contained within the multiframe is not needed. Therefore, the M1 and M2 bitstreams are used to convey information that is unique to non-transparent data transmission, as indicated below:

- The M1 bitstream is used to indicate which RLP frame half is contained within the 288 bit data block currently being transmitted.
- In the downlink direction, the M2 bitstream is used for DTX indication. The M2 bit is spare in the uplink direction.

## 4.2 Non-Transparent Service

### 4.2.1 Different RLP Version Types

To accommodate 14.4 kbit/s transmission, a “double-RLP” frame structure has been developed for the 14.4 kbit/s services. The structure is based on the concatenation of two frames, with the removal of redundant header and FCS fields. The resulting structure is 576 bits long, and has an information field 520-528 bits long (depending on the RLP version used). The frame repeats every 40 msec, yielding a radio interface user rate of 13.0 - 13.2 kbit/s. As shown below, the 14.4 kbit/s frame structure has been introduced into each version of RLP. This was done to prevent tying the implementation of 14.4 kbit/s services to the deployment of HSCSD services (HSCSD service requires the use of RLP version-2).

#### RLP Versions 0 & 1:

##### **RLP v0/1, Basic Data Services (9.6 kbit/s and below):**

Figure 9a shows the structure of RLP versions 0 & 1 used for basic data services (9.6 kbit/s and below). The RLP frame is 240-bits long, consisting of a 16-bit header field, 8-bit L2R status field, 192-bit information field, and a 24-bit FCS field. The frame repeats every 20 msec, and with a 192-bit information field, yields a 9.6 kbit/s radio interface user rate. The difference between the two versions is the support for V.42*b*/s compression in version 1.

Header	Status	Information	FCS
16bits	8 bits	192 bits	24 bits

**Figure 9a - RLP Version 0/1 (basic data services – 9.6 kbit/s and below)**

##### **RLP v0/1, 14.4 kbit/s Data Services:**

Figure 9b shows the structure of RLP versions 0 & 1 used for 14.4 kbit/s data services. The “double-RLP” frame is 576-bits long, consisting of a 16-bit header field, 8-bit L2R status field, 528-bit information field, and a 24-bit FCS field. The frame repeats every 40 msec, and with a 528-bit information field, yields a 13.2 kbit/s radio interface user rate.

Header	Status	Information	FCS
16bits	8 bits	528 bits	24 bits

**Figure 9b - RLP Version 0/1 (14.4 kbit/s data services)**

#### RLP Version-2: Version 2 is used for multislot data services (HSCSD).

##### **RLP v2, Basic Data Services (9.6 kbit/s and below):**

Version-2 for basic data services is still 240 bits long, but increases the header from 16 bits to 24 bits to accommodate multislot connections. In order to minimize the effect of the increased frame header length on the radio interface user rate, the L2R status procedures were changed for version 2 – the status field is used for information unless an actual change of status has occurred.

**GSM 10.14 version 1.0.1: July 1997**

This means the radio interface user rate varies between 9.2 and 9.6 kbit/s (in practice it is closer to 9.6 kbit/s, since status changes occur infrequently). See figure 9c below.

Header	Status	Information	FCS
24bits	0 - 8 bits	184/192 bits	24 bits

**Figure 9c - RLP Version 2 (basic data services – 9.6 kbit/s and below)**

**RLP v2, 14.4 kbit/s Data Services:**

Figure 9d shows the structure of RLP version 2 for 14.4 kbit/s data services. The overall frame length is 576 bits. Depending on whether or not status information is transmitted (as explained above), the information field varies from 520 to 528-bits, yielding a radio interface user rate of 13.0 - 13.2 kbit/s (in practice, closer to 13.2 kbit/s).

Header	Status	Information	FCS
24bits	0 - 8 bits	520/528 bits	24 bits

**Figure 9d - RLP Version 2 (14.4 kbit/s data services)**

**4.2.1.1 Delay Analysis for 14.4 kbit/s RLP Frame Structure:**

The increased frame length used in the 14.4 kbit/s RLP frame structure causes an additional 20msec of delay in the interleaving process. This increases the roundtrip delay by approximately 40msec compared to current 9.6 data services. In addition, as noted earlier, a delay of  $\approx 400$  msec for the mobile end, and  $\approx 600$  msec for the network end (not additive) is incurred to reset the RLP frame structure whenever there is a handover between 14.4 kbit/s services, and basic data services. The reset procedure is described below in section 4.2.2.

**4.2.2 Resetting the RLP Structure – “REMAP” Procedure**

During a handover between 14.4 kbit/s service and a non-14.4 service, the RLP frame structure must be reset. This is referred to as a REMAP procedure – any frame that has not been correctly received must be remapped into the new RLP format and retransmitted. Following is a description of the REMAP procedure:

- When the channel coding changes between basic data service and 14.4 (e.g., due to a handover), both ends of the RLP connection are informed about it through an external event (signalling at the mobile-end, change of layer 1 framing at the network-end). Upon detection of the change in channel coding, each end of the connection enters a synchronization state.
- Upon entering the synchronization state, the RLP timers are halted, the transmission and reception windows are cleared, and all out-of-sequence frames are discarded.
- The mobile-end then begins sending “REMAP” RLP frames to the network-end. These frames indicate the N(R) number of the frame, in the previous RLP format, from which the network-end should remap the information into the new RLP format for transmission. The mobile-end will continue to send REMAP frames until it receives a REMAP-acknowledgement frame from the network-end.
- When the network-end receives a REMAP frame from the mobile-end, it acknowledges by sending a REMAP-acknowledgement frame (a REMAP frame with the command/response bit set to “response”) to the mobile-end. The information field of the REMAP-acknowledgement frame contains the N(R) number of the frame, in the previous RLP format, from which the mobile-end should remap the information into the new RLP format for transmission. The network-end responds with acknowledgement frames to all REMAP commands it receives from the mobile-end.
- The REMAP procedure is exited when the mobile-end receives a REMAP-acknowledgement frame from the network-end. Duplicate acknowledgement frames received by the mobile-end are discarded. After the REMAP-procedure is completed, the RLP-entities leave the synchronisation state and normal operation is resumed. On resuming normal operation, the transmission and reception

windows are emptied, and the N(S) numbering resumes from the value indicated in the REMAP messages by the N(R) number.

- In addition to the N(R) information contained in the REMAP frame from the mobile-end, any XID parameters that may need to be renegotiated for the new channel coding may be included within the information field of the REMAP frame. XID acknowledgement will be contained within the REMAP-acknowledgement frames from the network-end.

#### 4.2.3 Modifications to L2R Addressing

Because the frame length has been extended for 14.4, a modification to the L2R addressing procedures is needed to accommodate the situation where two status octets need to be transmitted (an infrequent occurrence, but one that can happen).

The current definition of the L2R status octet contains a 5-bit address field that gives the location of any second status octet in the frame. Accounting for bits that are currently reserved, a second status octet can be indicated if it is no further than 27 octets away in the frame. This is not an issue with the RLP frame used for basic data services (the information field is no more than 24 octets long), but it is an issue for 14.4 because the information field has been extended to 66 octets.

To address this problem, the 14.4 kbit/s services use the reserved address '27' in the address field of the L2R octet. A value of '27' in the address field indicates that the next status octet is more than 23 octets away, and thus the status information will be given in a two octet status field, instead of the normal single octet. The address value of '27' indicates that the following octet contains a displacement address indicating the position of the next status octet. The 7-bit address field in the second octet allows for a displacement of up to 127 octets (a displacement up to 64 octets is needed for 14.4). The 8<sup>th</sup> bit in octet 2 is left as a reserved bit. Refer to figure 10 below.

Octet1	SA	SB	X	1	1	0	1	1
Octet2	RES	A	A	A	A	A	A	A

**Figure 10 - Double-octet Status**

In practice, the need for double-octet status will be marginal – status octets are rarely needed in the first place, and a double-octet is only used when the displacement address is  $\geq 23$ .

## 4.2.4 Analysis of Asynchronous Non-Transparent Data Rates

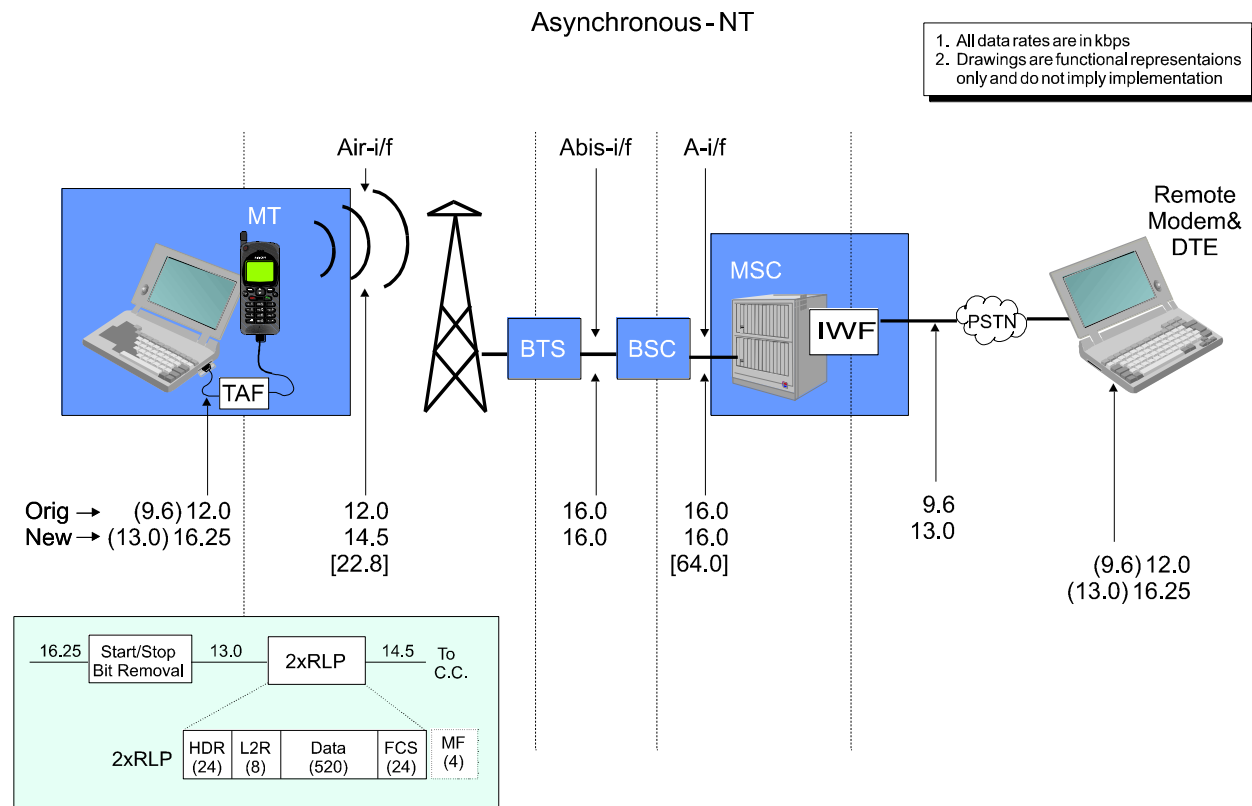


Figure 11 - Asynchronous NT Data Rates

The following text summarizes the 14.4 kbit/s asynchronous non-transparent data transfer scenario depicted in figure 11 above. For simplicity, the text is only written with respect to an outgoing data transmission from the MT.

**DTE-TAF interface:** The DTE sends user information to the TAF in 8-bit characters. Around each character, start and stop bits are included to delimit the information. The sum of the user's information (at 13 kbit/s) and the start/stop overhead bits (at 3.25 kbit/s) yields an overall terminal data rate of 16.25 kbit/s.

**Terminal Adaptation Function:** Once the incoming data stream enters the TAF, the start and stop bits are removed. Removal of these bits is a standard operation for non-transparent data – start/stop bits are not needed because bit positions within the RLP frame provide the necessary demarcation boundaries for characters. Once the start/stop overhead is removed, the user's information is inserted into the double-RLP frame structure described previously. The combination of the user's information at 13 kbit/s, and the RLP overhead, yields a user data rate of 14.4 kbit/s. Four multiframe bits are then appended to the data stream before it is sent to the MS for channel coding. For NT data, the multiframe bits are used to distinguish RLP frame halves, and for downlink DTX indication. The addition of the multiframe bits increases the radio interface rate to 14.5 kbit/s, as shown in the figure.

**Mobile Terminal and Radio Interface:** The data stream from the TAF enters the MT where it is channel coded as described in section 3. The output of the channel coding is a 22.8 kbit/s data stream that is sent over the radio interface.

**BTS:** Once the 22.8 kbit/s data stream is received from the radio interface, the BTS performs the forward error correction channel decoding procedures. The resulting 14.5 kbit/s data stream consists of the multiframe bits, and the RLP frame. The data is then packaged into the Extended-TRAU (E-TRAU) frame format described in Appendix A. The combination of the input data stream and the E-TRAU frame overhead yields a 16 kbit/s output on the *Abis*-interface.

Abis-Interface: The 16 kbit/s E-TRAU frame is sent over the *Abis*-interface to the BSC. See Appendix A for a description of the E-TRAU frame format and procedures.

TRAU: The TRAU receives the 16 kbit/s E-TRAU frame from the *Abis*-interface, and reformats the information into the A-TRAU format, described in Appendix B, for transmission on the A-interface. Reformatting of the data includes performing the Zero Sequence Positioning (ZSP) procedure described in the appendix.

A-Interface: Transmission on the A-interface is at 64 kbit/s. Just prior to transmitting the A-TRAU frame, fill bits are added by the TRAU to increase the data rate from 16 kbit/s to 64 kbit/s.

Note: Fill bits are used for non-HSCSD operation. For HSCSD a different procedure takes place. The A-interface for HSCSD can accommodate up to four (4) 16 kbit/s A-TRAU frames from one user. These frames are multiplexed together for transmission over the 64 kbit/s link. If a user has less than four channels in operation, fill bits are added to bring the total transmission rate up to 64 kbit/s.

MSC and IWF: The MSC receives the A-TRAU frame and routes it to the IWF. Upon reception, the IWF disassembles the A-TRAU frame and extracts the multiframe bits and the RLP frame data. The IWF then determines if there are any uncorrected errors in the RLP frames, and if so, uses the ARQ mechanisms in the radio link protocol to request retransmission of the appropriate frames from the TAF. The output from the IWF is the (error-free) user's information at 13 kbit/s. This data is routed out over the PSTN to the destination modem.

Destination Modem and Remote DTE: Upon reception of the incoming 13 kbit/s data stream, the destination modem inserts the start/stop overhead bits around each character. The resultant data stream, at 16.25 kbit/s, is then forwarded to the remote DTE.

4.2.5 Analysis of Synchronous Non-transparent Data Rates

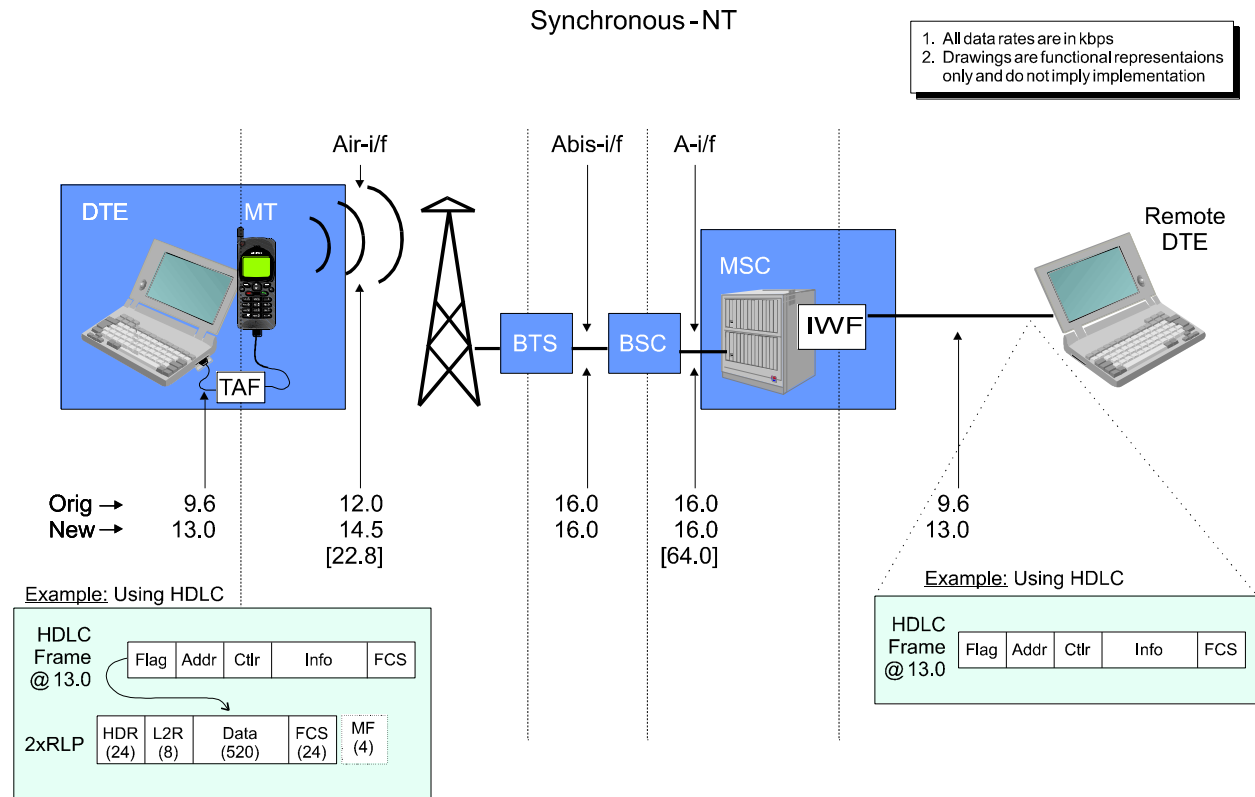


Figure 12 - Synchronous NT Data Rates

The following text summarizes the 14.4 kbit/s synchronous non-transparent data transfer scenario depicted in figure 12 above. For simplicity, the text is only written with respect to an outgoing data transmission from the MT.

**DTE-TAF interface:** The DTE sends information to the TAF at 13 kbit/s.

**Terminal Adaptation Function:** The TAF fills the double-RLP's information field as shown in the figure. Four multiframe bits are then appended to the data stream before it is sent to the MS for channel coding. For NT data, the multiframe bits are used to distinguish RLP frame halves, and for downlink DTX indication. The addition of the multiframe bits increases the radio interface rate to 14.5 kbit/s, as shown in the figure.

**Mobile Terminal and Radio Interface:** The data stream from the TAF enters the MT where it is channel coded as described in section 3. The output of the channel coding is a 22.8 kbit/s data stream that is sent over the radio interface.

**BTS:** Once the 22.8 kbit/s data stream is received from the radio interface, the BTS performs the forward error correction channel decoding procedures. The resulting 14.5 kbit/s data stream consists of the multiframe bits, and the RLP frame. The data is then packaged into the Extended-TRAU (E-TRAU) frame format described in Appendix A. The combination of the input data stream and the E-TRAU frame overhead yields a 16 kbit/s output on the *Abis*-interface.

**Abis-Interface:** The 16 kbit/s E-TRAU frame is sent over the *Abis*-interface to the BSC. See Appendix A for a description of the E-TRAU frame format and procedures.

**TRAU:** The TRAU receives the 16 kbit/s E-TRAU frame from the *Abis*-interface, and reformats the information into the A-TRAU format, described in Appendix B, for transmission on the A-interface. Reformatting of the data includes performing the Zero Sequence Positioning (ZSP) procedure described in the appendix.

A-Interface: Transmission on the A-interface is at 64 kbit/s. Just prior to transmitting the A-TRAU frame, fill bits are added by the TRAU to increase the data rate from 16 kbit/s to 64 kbit/s.

Note: Fill bits are used for non-HSCSD operation. For HSCSD a different procedure takes place. The A-interface for HSCSD can accommodate up to four (4) 16 kbit/s A-TRAU frames from one user. These frames are multiplexed together for transmission over the 64 kbit/s link. If a user has less than four channels in operation, fill bits are added to bring the total transmission rate up to 64 kbit/s.

MSC and IWF: The MSC receives the A-TRAU frame and routes it to the IWF. Upon reception, the IWF disassembles the A-TRAU frame and extracts the multiframe bits and the RLP frame data. The IWF then determines if there are any uncorrected errors in the RLP frames, and if so, uses the ARQ mechanisms in the radio link protocol to request retransmission of the appropriate frames from the TAF. The output is the (error-free) user's information. The resultant data stream, at 13 kbit/s, is then sent out from the IWF to the remote DTE.

Remote DTE: The remote DTE accepts the incoming data at 13 kbit/s.



## 4.3 Transparent Service

### 4.3.1 Synchronous Transparent Data

The GSM 14.4 kbit/s data services support protocol-independent data transmission with a radio interface user rate of 14.4 kbit/s.

14.4 kbit/s data transmission is achieved by replacing the 8 status bits in the 72-bit V.110 frame with data (creating a so-called "14.4-Tdata" frame). The 14.4-Tdata frame repeats every 5msec, yielding a user data rate of 14.4 kbit/s. The status information is transported using the multiframe structure described in section 4.1.

Note: Only two of the 8 status bits are needed – the other 6 bits are redundant.

## 4.3.1.2 Analysis of Synchronous Transparent Data Rates

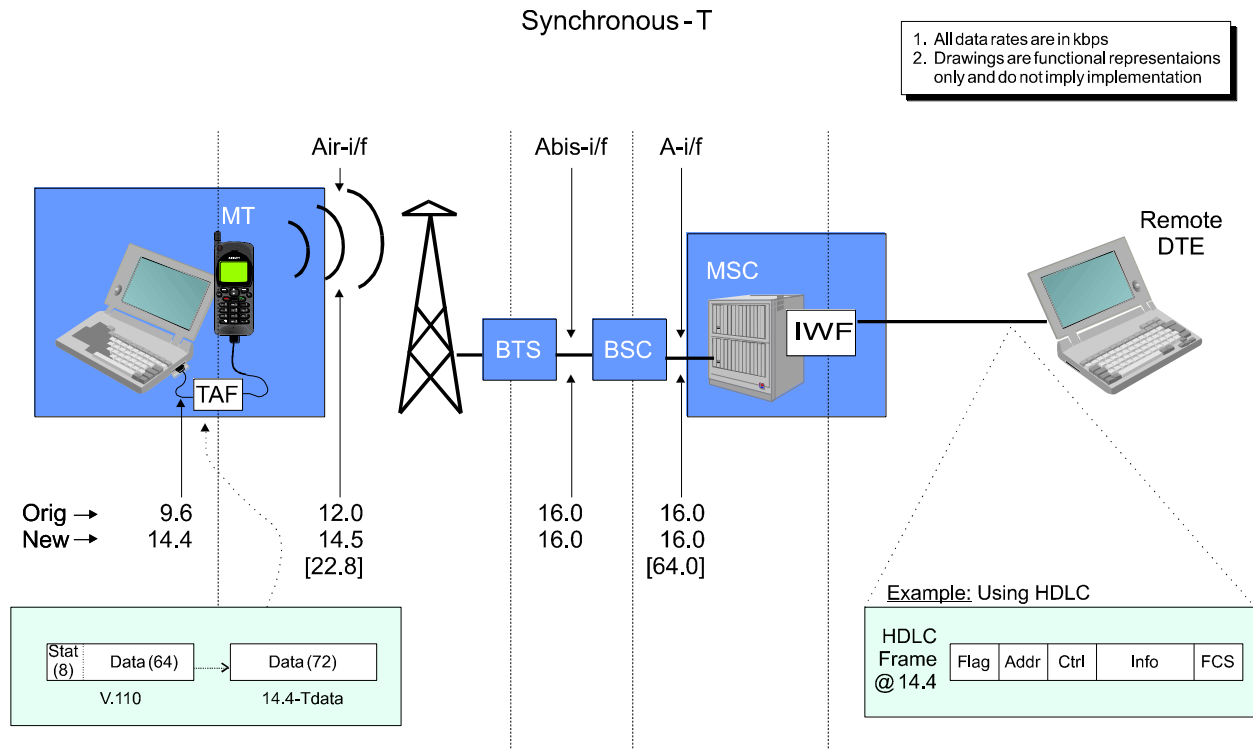


Figure 13 - Synchronous Transparent Data Rates

The following text summarizes the 14.4 kbit/s synchronous transparent data transfer scenario depicted in figure 13 above. For simplicity, the text is only written with respect to an outgoing data transmission from the MT.

**DTE-TAF interface:** The DTE sends information to the TAF at 14.4 kbit/s.

**Terminal Adaptation Function:** The TAF accepts the incoming data and inserts it into the 14.4-Tdata frames. Multiframing information is then appended, and the 14.5 kbit/s data stream is sent from the TAF to the MS.

**Mobile Terminal and Radio Interface:** The data stream from the TAF enters the MT where it is channel coded as described in section 3. The output of the channel coding is a 22.8 kbit/s data stream that is sent over the radio interface.

**BTS:** Once the 22.8 kbit/s data stream is received from the radio interface, the BTS performs the forward error correction channel decoding procedures. The resulting 14.5 kbit/s data stream consists of the multiframing information, and the user's data. The data is then packaged into the Extended-TRAU (E-TRAU) frame format described in Appendix A. The combination of the input data stream and the E-TRAU frame overhead yields a 16 kbit/s output on the *Abis*-interface.

**Abis-Interface:** The 16 kbit/s E-TRAU frame is sent over the *Abis*-interface to the BSC. See Appendix A for a description of the E-TRAU frame format and procedures.

**TRAU:** The TRAU receives the 16 kbit/s E-TRAU frame from the *Abis*-interface, and reformats the information into the A-TRAU format, described in Appendix B, for transmission on the A-interface. Reformatting of the data includes performing the Zero Sequence Positioning (ZSP) procedure described in the appendix.

**A-Interface:** Transmission on the A-interface is at 64 kbit/s. Just prior to transmitting the A-TRAU frame, fill bits are added by the TRAU to increase the data rate from 16 kbit/s to 64 kbit/s.

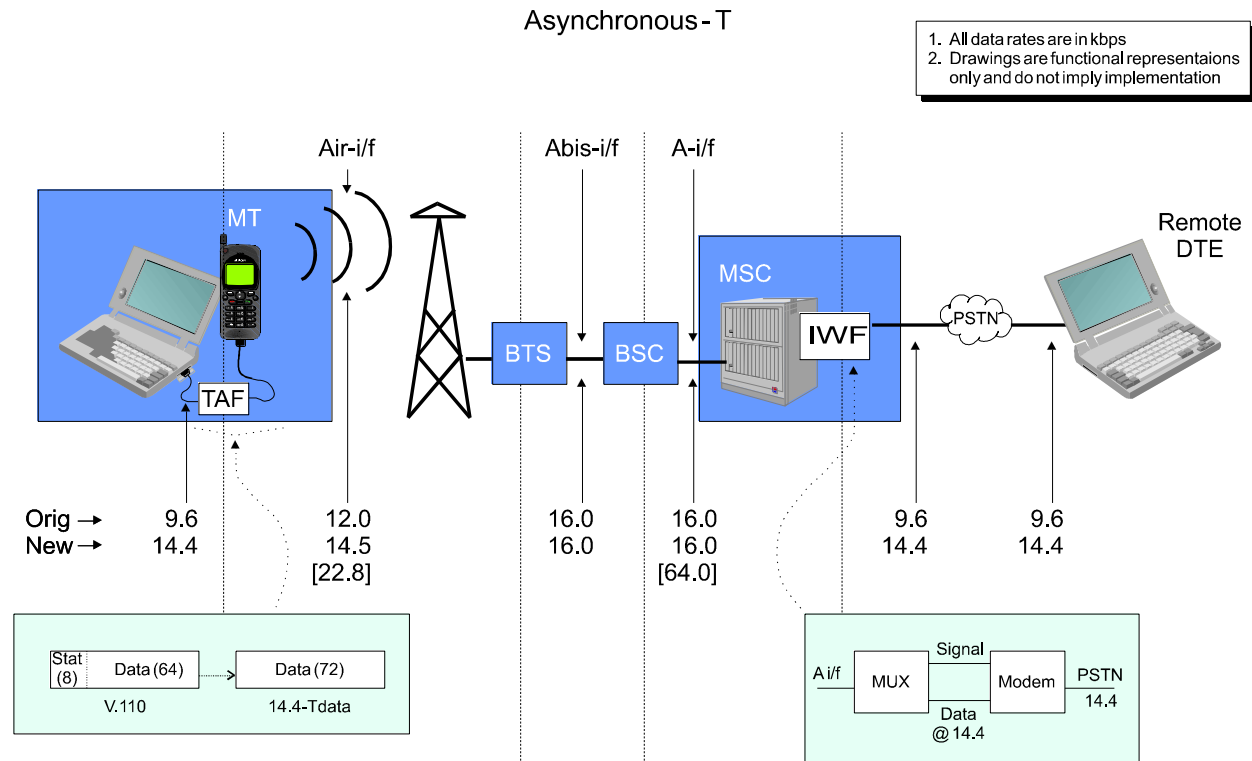
Note: Fill bits are used for non-HSCSD operation. For HSCSD a different procedure takes place. The A-interface for HSCSD can accommodate up to four (4) 16 kbit/s A-TRAU frames from one user. These frames are multiplexed together for transmission over the 64 kbit/s link. If a user has less than four channels in operation, fill bits are added to bring the total transmission rate up to 64 kbit/s.

MSC and IWF: The MSC receives the A-TRAU frame and routes it to the IWF. Upon reception, the IWF disassembles the A-TRAU frame, extracts the multiframe information and the user's information. The data stream is then sent at 14.4 kbit/s from the IWF out to the remote DTE.

Remote DTE: The remote DTE accepts the incoming data at 14.4 kbit/s.

### 4.3.2 Asynchronous Transparent Data

Asynchronous data includes Start- and Stop-bits indicating the beginning and end of the actual "user" data characters. Since data transfer over GSM is synchronous, the arriving asynchronous data flow must be mapped into synchronous data frames. For 14.4 asynchronous transparent data service, the data stream is mapped to 14.4-Tdata frames as described in the preceding section.



The following text summarizes the 14.4 kbit/s asynchronous transparent data transfer scenario depicted in figure 14 above. For simplicity, the text is only written with respect to an outgoing data transmission from the MT.

**DTE-TAF interface:** The DTE sends user information to the TAF in 8-bit characters. Around each character, start and stop bits are included to delimit the information. The sum of the user's information and the start/stop overhead bits combine to yield an overall data rate of 14.4 kbit/s.

**Terminal Adaptation Function:** The TAF accepts the incoming data and inserts it into the 14.4-Tdata frames. Multiframing information is then appended, and the 14.5 kbit/s data stream is sent from the TAF to the MS.

**Mobile Terminal and Radio Interface:** The data from the TAF enters the MT where it is channel coded as described in section 3. The output of the channel coding is a 22.8 kbit/s data stream that is sent over the radio interface.

**BTS:** Once the 22.8 kbit/s data stream is received from the radio interface, the BTS performs the forward error correction channel decoding procedures. The resulting 14.5 kbit/s data stream consists of the multiframing information, and the user's data. The data is then packaged into the Extended-TRAU (E-TRAU) frame format described in Appendix A. The combination of the input data stream and the E-TRAU frame overhead yields a 16 kbit/s output on the *Abis*-interface.

**Abis-Interface:** The 16 kbit/s E-TRAU frame is sent over the *Abis*-interface to the BSC. See Appendix A for a description of the E-TRAU frame format and procedures.

**TRAU:** The TRAU receives the 16 kbit/s E-TRAU frame from the *Abis*-interface, and reformats the information into the A-TRAU format, described in Appendix B, for transmission on the A-interface. Reformatting of the data includes performing the Zero Sequence Positioning (ZSP) procedure described in the appendix.

**A-Interface:** Transmission on the A-interface is at 64 kbit/s. Just prior to transmitting the A-TRAU frame, fill bits are added by the TRAU to increase the data rate from 16 kbit/s to 64 kbit/s.

Note: Fill bits are used for non-HSCSD operation. For HSCSD a different procedure takes place. The A-interface for HSCSD can accommodate up to four (4) 16 kbit/s A-TRAU frames from one user. These frames are multiplexed together for transmission over the 64 kbit/s link. If a user has less than four channels in operation, fill bits are added to bring the total transmission rate up to 64 kbit/s.

MSC and IWF: The MSC receives the A-TRAU frame and routes it to the IWF. Upon reception, the IWF disassembles the A-TRAU frame, and extracts the multiframe information and the asynchronous data. The data is then sent to the IWF's modem pool. The modem pool accepts the data and routes it over the PSTN to the destination modem and remote DTE.

Destination Modem and Remote DTE: Upon reception of the incoming data stream, the destination modem inserts the start/stop overhead bits around each character. The sum of the user's information and the start/stop overhead bits combine to yield an overall data rate of 14.4 kbit/s.

**4.3.3 Facsimile Service**

Facsimile service is a significant feature of GSM, thus support for 14.4 kbit/s fax is an important benefit of the new 14.4 kbit/s data service.

Fax can operate over either transparent or non-transparent data services. However, facsimile over non-transparent service has not been implemented in GSM. Therefore, this document only addresses facsimile transmissions over transparent data channels, using the T.30 facsimile protocol. Facsimile transmission has two main operating modes: binary coded signalling (BCS) and high speed data transfer. BCS uses frequency shift keying (FSK) and has a data rate of 300 bps. The T.30 fax protocol operates on page boundaries, and switches between BCS for inter-page signalling, and higher data rates (2400 to 14.4 kbit/s) during actual page transmission.

When sending or receiving facsimile data in either error correct mode (ECM) or normal fax data mode (NFD), GSM's V.110 status and end-to-end bits (S, X and E) are not needed (the V.24 signals do not change during page transfer, and the end-to-end data rate is defined at call setup). Therefore, the V.110 frame (data + status) is effectively replaced with a 72-bit unstructured field that can be filled with data (a "14.4-fax" frame). This 72 bit field repeats every 5 msec yielding a total data rate of 14.4 kbit/s that may be completely used for facsimile data.

## 4.3.3.1 Analysis of Facsimile Data Rates

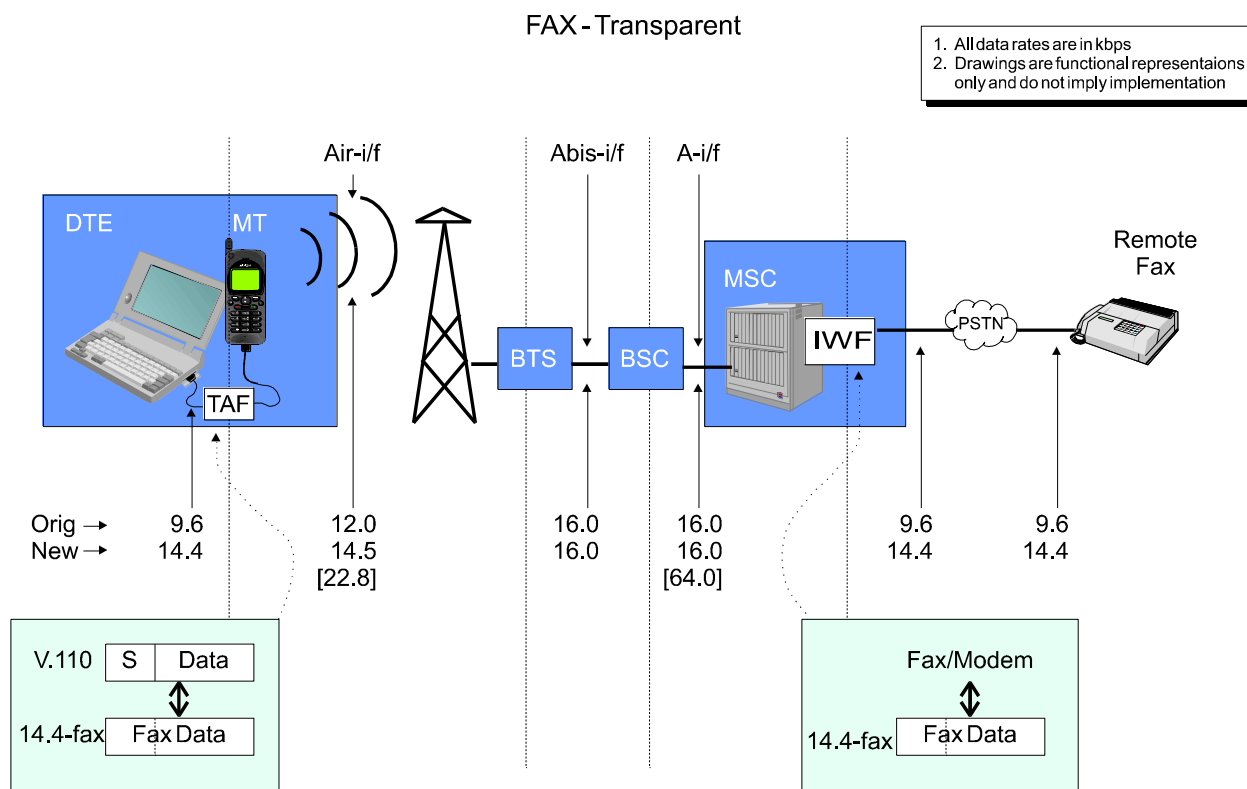


Figure 15 - Synchronous Transparent Facsimile Data Rates

The following text summarizes the 14.4 kbit/s synchronous transparent facsimile transfer scenario depicted in figure 15 above. For simplicity, the text is only written with respect to an outgoing facsimile transmission from the MT.

**DTE-TAF interface:** The facsimile information is sent from the DTE to the TAF format at 14.4 kbit/s (in HDLC format for ECM facsimile, and as unstructured data for NFD mode).

**Terminal Adaptation Function:** The breakout box for the TAF in the diagram above refers to the replacement of the V.110 status information with image data (as described in the text in the preceding section). The result is a 72-bit 14.4-fax frame that repeats every 5 msec, yielding an overall data rate of 14.4 kbit/s. The TAF fills the 14.4-fax frame with the information from the DTE, appends the multiframe bits, and forwards the 14.5 kbit/s data stream to the MT. The multiframe bits are not actually used for facsimile services, however the bits must be appended in order to keep the channel coding scheme common for all 14.4 kbit/s data services.

**Mobile Terminal and Radio Interface:** The data from the TAF enters the MT where it is channel coded as described in section 3. The output of the channel coding is a 22.8 kbit/s data stream that is sent over the radio interface.

**BTS:** Once the 22.8 kbit/s data stream is received from the radio interface, the BTS performs the forward error correction channel decoding procedures. The resulting 14.5 kbit/s data stream consists of the multiframe information, and the facsimile data. The data is then packaged into the Extended-TRAU (E-TRAU) frame format described in Appendix A. The combination of the input data stream and the E-TRAU frame overhead yields a 16 kbit/s output on the *Abis*-interface.

**Abis-Interface:** The 16 kbit/s E-TRAU frame is sent over the *Abis*-interface to the BSC. See Appendix A for a description of the E-TRAU frame format and procedures.

**TRAU:** The TRAU receives the 16 kbit/s E-TRAU frame from the *Abis*-interface, and reformats the information into the A-TRAU format, described in Appendix B, for transmission on the *A*-interface. Reformatting of the data includes performing the Zero Sequence Positioning (ZSP) procedure described in the appendix.



A-Interface: Transmission on the A-interface is at 64 kbit/s. Just prior to transmitting the A-TRAU frame, fill bits are added by the TRAU to increase the data rate from 16 kbit/s to 64 kbit/s.

Note: Fill bits are used for non-HSCSD operation. For HSCSD a different procedure takes place. The A-interface for HSCSD can accommodate up to four (4) 16 kbit/s A-TRAU frames from one user. These frames are multiplexed together for transmission over the 64 kbit/s link. If a user has less than four channels in operation, fill bits are added to bring the total transmission rate up to 64 kbit/s.

MSC and IWF: The MSC receives the A-TRAU frame and routes it to the IWF. Upon reception, the IWF disassembles the A-TRAU frame, extracts the multiframe information and the facsimile data, and forwards the information to the IWF's modem pool. The modem pool accepts the information and routes it over the PSTN to the destination fax machine.

Destination Fax Machine: The destination fax machine receives the incoming information at 14.4 kbit/s (in HDLC format for ECM facsimile, and as unstructured data for NFD mode).

## **Annex A: Structure of the Extended-TRAU (E-TRAU) Frame Format used for 14.4 kbit/s Data Transmission on the *Abis*-Interface**

For 14.4 kbit/s services, an Extended-TRAU frame format is used to transmit data across the *Abis*-interface. The structure of the E-TRAU frame is based on the current TRAU frame used on the *Abis*-interface for basic data services.

The TRAU frame used for basic data services does not have enough room in the payload to transmit the 14.5 kbit/s data stream (14.4 kbit/s payload + 100 bps multiframe) across the *Abis*-interface. To accommodate the increased data rates, two frame formats are used for 14.4 kbit/s services. The first frame is referred to as a "14.4 Data TRAU Frame", and is used for synchronizing the *Abis*-interface. The second frame format is referred to as a "14.4 Extended-Data TRAU Frame", and is used for transferring data and multiframe information.

### **A.1 Synchronization of the *Abis*-Interface for 14.4 kbit/s Services**

The synchronization of the *Abis*-interface is carried out in a master-slave handshaking arrangement that is controlled by the BTS. The BTS begins the synchronization procedure by sending the 14.4 Data TRAU Frames shown in Figure A1 to the TRAU. 14.4 Data TRAU Frames are identified by setting the frame type code to '10100' in bits C1-C5. Upon receipt of a 14.4 Data TRAU Frame, the TRAU acknowledges by sending 14.4 Data TRAU Frames back to the BTS on the downlink. The BTS will continue to send 14.4 Data TRAU Frames to the TRAU until it receives an acknowledgement frame from the TRAU.

Synchronization of the *Abis*-interface is necessary at initial setup. Resynchronization is necessary after a handover, and whenever synchronization is lost during data transfer (an unusual occurrence).

The control and payload bits in 14.4 Data TRAU Frames are ignored (except for bits C1-C5). The same frame structure is used for both transparent and non-transparent data services.

Octet no.	Bit number							
	1	2	3	4	5	6	7	8
0	0	0	0	0	0	0	0	0
1	0	0	0	0	0	0	0	0
2	1	C1	C2	C3	C4	C5	C6	C7
3	C8	C9	C10	C11	C12	C13	C14	C15
4	1	Data frame position 1 63 bits. (72 bits including bit position 1)						.
5	1							.
6	1							.
7	1							.
8	1							.
9	1							.
10	1							.
11	1	.						
12	1	.						
13	1	Data frame position 2						.
14	1							.
15	1							.
16	1							.
17	1							.
18	1							.
19	1							.
20	1	.						
21	1	.						
22	1	Data frame position 3						.
23	1							.
24	1							.
25	1							.
26	1							.
27	1							.
28	1							.
29	1	.						
30	1	.						
31	1	Data frame position 4						.
32	1							.
33	1							.
34	1							.
35	1							.
36	1							.
37	1							.
38	1	.						
39	1	.						

Figure A1 – 14.4 Data TRAU Frame

## A.2 14.4 kbit/s Services Data Transmission on the *Abis*-Interface

When the BTS receives an acknowledgement 14.4 Data TRAU Frame from the TRAU, it knows that synchronization has been achieved on the *Abis*-interface. At this point, the BTS will exit the synchronization phase, and begin data transmission on the *Abis*-interface. For data transmission, the 14.4 Extended Data TRAU Frame shown in Figure A2 is used. Extended Data TRAU Frames are identified by setting the frame type code to '11111' in bits C1-C5.

Upon receipt of a 14.4 Extended Data TRAU Frame, the TRAU transitions its mode of operation to data transmission, and also acknowledges by sending 14.4 Extended Data TRAU Frames back to the BTS on the downlink. At this point, both the uplink and downlink are fully synchronized and data transmission is occurring.

The main difference between the 14.4 Extended Data TRAU frame structure, and the standard TRAU frame used for basic data services, is the absence of embedded synchronization zeroes in the payload. The synchronization zeroes have been removed in order to make room for the 14.5 kbit/s data stream. The zeroes may be removed because the synchronization of the interface has already been established by the handshaking procedure described in the preceding section. The resulting frame structure contains 288 data payload bits, 2 multiframe bits (M1, M2), and 30 control and synchronization bits. The frame repeats every 20 msec yielding an overall data rate of 16 kbit/s for the frame, and 14.5 kbit/s for the data payload + multiframe. The same frame structure is used for both transparent and non-transparent data services.

Bit C6 is used in the uplink for idle/data frame indication. This bit is normally set to indicate 'data', which means that the information in the frame is valid. However, whenever radio interface synchronization is lost (e.g., during a handover), this bit is set to 'idle' to indicate the data is not valid. This allows the IWF to discard the data further upstream. Bit C6 is used in the downlink to indicate an Uplink Frame Error (UFE) condition. This bit is set by the TRAU to inform the BTS that synchronization is not being maintained over the *Abis*-interface. The UFE bit is integrated by the BTS – if enough consecutive UFE indications are received, a resynchronization is triggered, and the procedures described in the previous section are carried out to regain synchronization.

Bits C7-C15 are spare.

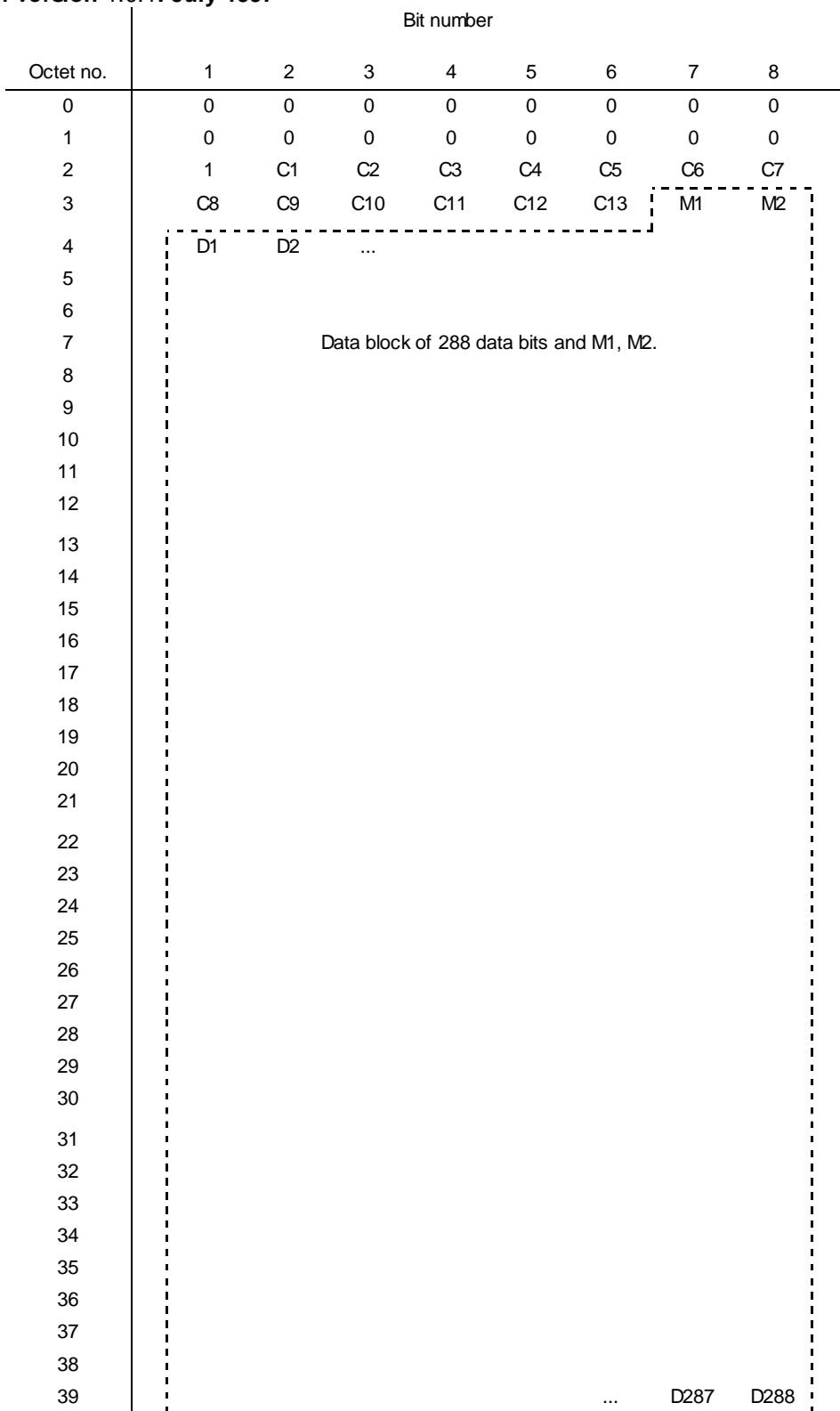


Figure A2 – 14.4 Extended Data TRAU Frame

## Annex B: Structure of the A-TRAU Frame Format used for 14.4 kbit/s Data Transmission on the A-Interface

For 14.4 kbit/s services, the A-TRAU frame format shown in Figure B1 is used to transmit data across the A-interface.

0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
1	C1	C2	C3	C4	C5	M1	M2
Z1	D1	D2	D3	D4	D5	D6	D7
D8	D9	D10	D11	D12	D13	D14	D15
D16	D17	D18	D19	D20	D21	D22	D23
D24	D25	D26	D27	D28	D29	D30	D31
D32	D33	D34	D35	D36	Z2	D1	D2
D3	D4	D5	D6	D7	D8	D9	D10
D11	D12	D13	D14	D15	D16	D17	D18
D19	D20	D21	D22	D23	D24	D25	D26
D27	D28	D29	D30	D31	D32	D33	D34
D35	D36	Z3	D1	D2	D3	D4	D5
D6	D7	D8	D9	D10	D11	D12	D13
D14	D15	D16	D17	D18	D19	D20	D21
D22	D23	D24	D25	D26	D27	D28	D29
D30	D31	D32	D33	D34	D35	D36	Z4
D1	D2	D3	D4	D5	D6	D7	D8
D9	D10	D11	D12	D13	D14	D15	D16
D17	D18	D19	D20	D21	D22	D23	D24
D25	D26	D27	D28	D29	D30	D31	D32
D33	D34	D35	D36	Z5	D1	D2	D3
D4	D5	D6	D7	D8	D9	D10	D11
D12	D13	D14	D15	D16	D17	D18	D19
D20	D21	D22	D23	D24	D25	D26	D27
D28	D29	D30	D31	D32	D33	D34	D35
D36	Z6	D1	D2	D3	D4	D5	D6

D7	D8	D9	D10	D11	D12	D13	D14
D15	D16	D17	D18	D19	D20	D21	D22
D23	D24	D25	D26	D27	D28	D29	D30
D31	D32	D33	D34	D35	D36	<b>Z7</b>	D1
D2	D3	D4	D5	D6	D7	D8	D9
D10	D11	D12	D13	D14	D15	D16	D17
D18	D19	D20	D21	D22	D23	D24	D25
D26	D27	D28	D29	D30	D31	D32	D33
D34	D35	D36	<b>Z8</b>	D1	D2	D3	D4
D5	D6	D7	D8	D9	D10	D11	D12
D13	D14	D15	D16	D17	D18	D19	D20
D21	D22	D23	D24	D25	D26	D27	D28
D29	D30	D31	D32	D33	D34	D35	D36

Figure B1 – A-TRAU Frame

The structure of the A-TRAU frame has been designed to maintain the same frame length as the E-TRAU frame used on the *Abis*-interface. In addition, to guarantee data transparency, a Zero Sequence Position (ZSP) procedure is used for the data payload. Data transparency is especially important across the A-interface, because the A-interface is a multivendor environment. The ZSP procedure is described in section B.1.

Bits C1-C4 contain the bit pattern '0111' and are used to indicate the frame type code for 14.4 kbit/s data service.

Bit C5 is used for idle/data frame indication in the uplink direction, and for UFE indication in the downlink direction. This bit corresponds to the C6 bit described in Appendix A for the E-TRAU frame structure.

Bits M1 and M2 contain the multiframe bits described in section 4.1.

The remainder of the frame is used to transfer the 288-bit data payload, and 8 control bits (Z1-Z8) that are used as part of the ZSP procedure. The data payload has been broken down into 8 blocks – each 36-bit block is individually processed using the ZSP procedure. This has been done to minimize the amount of buffering and latency required to process the data.

## B.1 Zero Sequence Position (ZSP) Procedure

The ZSP procedure, as noted earlier, is used to guarantee data transparency. This is done by searching for strings of 8 zeroes, and replacing each of them with a special ZSP field. Removing any occurrence of 8 zeroes from the data payload guarantees that a false synchronization will not occur within the data payload (if 8 zeroes occur in the payload it would be because a handover had occurred). Following is a description and illustrative example of the ZSP procedure.

### B.1.1 ZSP Procedure Description

A Zero Sequence Position (ZSP) field is used to account for the occurrence of eight zeroes in the 36-bit data field. A sequence of eight zeroes is referred to as a "Z-sequence".

Note: A sequence of eight zeroes is considered as a block (e.g., a stream of eleven consecutive zeroes would produce only one ZSP, not four ZSP's).

The ZSP field is defined as shown in Figure B2 :

Bit #	1	2	3	4	5	6	7	8
Defn.	1	C	A0	A1	A2	A3	A4	1

**Figure B2 – ZSP Field**

Bit1/Bit8: Locking bit. Set to 1 to prevent the false occurrence of a Z-sequence.

Bit 2: Continuation bit. '0' means that there is another ZSP in the data field. '1' means that this is the last ZSP.

Bits 3-7: Address of the next Z-sequence (eight zeroes) to be inserted. The address '00001' corresponds to the bit D1; the address '11101' to the bit D29, (A0 is the msb, A4 is the lsb). Note that D29 is the highest address needed because it is the last place where a Z-sequence can be inserted before the end of the 36-bit block.

The ZSP procedure is applied in each of the eight 36-bit data blocks shown in Figure B1. Bit  $Z_i$  indicates whether a ZSP is used in the  $i^{\text{th}}$  36 bit data field ( $i=1$  to 8). The coding of the  $Z_i$  bit is as shown in Figure B3:

$Z_i$ ( $i=1..8$ )	Definition
1	No ZSP substitution in the data block
0	At least one ZSP substitution in the data block

**Figure B3 – ZSP Field**

If the  $Z_i$  bit indicates no substitution, the output data bits in the data block are equal to the input data bits. If the  $Z_i$  bit indicates at least one substitution, bits D1-D8 contain the first ZSP.

### B.1.2 ZSP Procedure Example

The following description indicates the general operating procedures for the ZSP procedure. It does not indicate a required implementation of the ZSP encoding procedure.



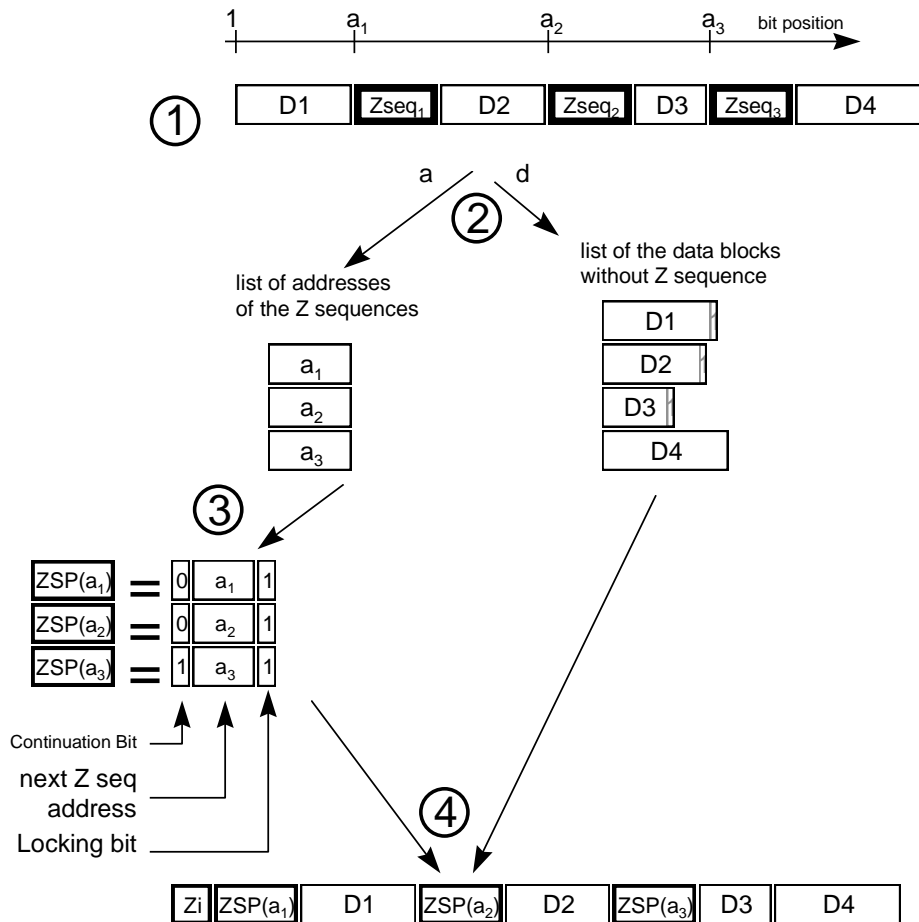


Figure B4 – ZSP Encoding Example

**Step 1 :**

The input 36-bit data block is considered as a bit stream in which the bits are numbered from 1 to 36.

This bit stream contains 0, 1 or several Z-sequences (Zseq<sub>1</sub> to Zseq<sub>3</sub> on the figure)

The Z-sequence is a sequence of 8 consecutive zeroes: '0000 0000'

**Step 2 :**

Starting from this bit stream, two lists are built up:

- **2-a :** the 'a' list which contains the address of the first bit of each Z-sequences.
- **2-d :** the 'd' list which contains all the data blocks which do not have the Z-sequence.

**Step 3 :**

The 'a' list is transformed to build the ZSP list. Each ZSP element is used to indicate:

- the address of the next Z-sequence
- if yet another ZSP element will be found at this address (link element)

**Step 4 :**

The output 37-bit sub-frame is built from:

- the  $Z_i$  field, which indicates whether the original message has been transformed or not. In the example given in figure B4,  $Z_i$  should be set to '0' to indicate that at least one substitution has occurred.
- the interleaving of the ZSP and D elements.

Since the ZSP elements have exactly the same length as the Z-sequence, the sub-frame length is only increased by one (the  $Z_i$  bit), no matter how many substitutions are made.

## Annex C: Approved Change Requests pertaining to 14.4 kbit/s Workitem

SMG#	TDoc	SPEC	VERS	CR	REV	PHASE	CAT	SUBJECT
s21	232/97	02.03	5.1.0	A003		2+	B	Support of 14.4 kbit/s (Radio interface related)
s21	232/97	02.06	5.0.0	A005		2+	B	Support of 14.4 kbit/s (Radio interface related)
s22	454/97	02.34	5.0.0	A007		2+	B	WI 14.4 kbit/s user data
s22	456/97	03.10	5.1.0	A005		2+	B	Introduction of 14.4 kbit/s
s22	460/97	03.45	5.1.0	A003	1	2+	B	Introduction of 14.4 kbit/s
s21	232/97	04.03	5.0.0	A003		2+	B	Support of 14.4 kbit/s (Radio interface related)
s22	425/97	04.03	5.0.0	A004		2+	F	14.4kbps data service
s22	425/97	04.08	5.3.0	A213		2+	F	Deletion of code points for 7.2 service
s21	232/97	04.21	5.0.0	A005		2+	B	Support of 14.4 kbit/s (Radio interface related)
s22	411/97	04.21	5.1.1	A006	4	2+	F	Corrections and improvements for 14.4 kbit/s
s22	411/97	04.22	5.1.0	A008	3	2+	F	14.4 kbit/s corrections and alignments
s21	232/97	04.22	5.0.1	A009		2+	B	Support of 14.4 kbit/s (Radio interface related)
s21	232/97	05.01	5.1.0	A008		2+	B	Support of 14.4 kbit/s (Radio interface related)
s21	232/97	05.02	5.2.0	A014		2+	B	Support of 14.4 kbit/s (Radio interface related)
s21	232/97	05.03	5.2.1	A010		2+	B	Support of 14.4 kbit/s (Radio interface related)
s21	232/97	05.05	5.3.0	A037		2+	B	Support of 14.4 kbit/s (Radio interface related)
s21	232/97	07.01	5.3.0	A020		2+	B	Support of 14.4 kbit/s (Radio interface related)
s22	411/97	07.01	5.4.0	A021		2+	B	Provide interworking at higher rates than 28.8 kbit/s (as 33.6 kbit/s).
s22	411/97	07.01	5.4.0	A022		2+	B	Correction 14.4 and synchronization
s21	232/97	07.02	5.1.1	A007		2+	B	Support of 14.4 kbit/s (Radio interface related)
s22	411/97	07.02	5.2.0	A009		2+	B	Corrections and improvements for 14.4 kbit/s
s21	232/97	07.03	5.0.1	A006		2+	B	Support of 14.4 kbit/s (Radio interface related)
s22	411/97	07.03	5.1.0	A007		2+	B	Corrections and improvements for 14.4 kbit/s
s22	458/97	08.08	5.4.0	A077		2+	B	WI 14.4 kbit/s user data
s22	544/97	08.08	5.4.0	A078	1	2+	F	Correction of channel type code points
s22	457/97	08.20	5.1.0	A003	2	2+	B	WI 14.4 kbit/s user data
s22	425/97	08.58	5.2.0	A021		2+	C	Deletion of 7.4 channel coding
s22	434/97	08.60	5.0.2	A006		2+	B	Introduction of 14.5 Kbit/s channel coding
s22	455/97	09.07	5.3.0	A028		2+	B	WI 14.4 kbit/s user data

<b>Document history</b>		
<b>Date</b>	<b>Version</b>	<b>Comments</b>
17 July 96	00	DRAFT - Initial release.
19 Aug 96	01	DRAFT - Major rewrite to all sections.
11 Sept 96	02	DRAFT - Rewrites and additions to all sections
30 Sept 96	T1P1.5/96-227R1	DRAFT - Version submitted to T1P1.5 Plenary after review on 17 Sept by 14.4 adHoc
2 Oct 96	T1P1.5/96-227R2	DRAFT - Minor Rewrites following review on October 1-2 by 14.4 adHoc
21 Oct 96	T1P1.5/96-227R3	DRAFT - Rewrites to several sections. Prepared for October 21-22 review by 14.4 adHoc.
18 Nov 96	T1P1.5/96-227R4	DRAFT - Rewrites following October 21 adHoc. R4 prepared for November 18-22 T1P1.5 meeting.
27 May 97	T1P1.5/97-234	Supersedes T1P1.5/96-227R4. Major rewrites to all sections to reflect changes in the design of the 14.4 kbit/s service, and to reflect the change in scope of the document to a system description. Prepared for May 27-30 T1P1.5 meeting.
28 May 97	T1P1.5/97-234R1	Review and minor updates by 14.4 adHoc group. R1 version submitted to T1P1.5 Plenary for liaison to ETSI, along with the final 14.4 CR package.
10/06/97	PT SMG PD 10.14v1.0.0	Converted to PT SMG permanent document. Approved at SMG#22.
23/07/97	PD 10.14v1.0.1	Correction of CR number 04.22 A009
Rapporteur: ETSI / PT12 (SMG).		
Email: stefan.aprath@etsi.fr                      Ph:+33.4.92 94 4324                      Fax:+33.4.93 65 28 17		