European ITS Framework Architecture

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Communication Architecture

Annex 2 – Details of ITS related Communications Technologies

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Executive Summary

This Document contains the second Annex to the Main part of European ITS Framework Architecture Deliverable Document D 3.3, which provides a description of the European ITS Communication Architecture. It provides additional and supporting information to the Main Document. This information comprises a description of the various technologies being currently used for system communications, an analysis of their main characteristics and a description of the OSI Model.

1 Introduction

1.1 Purpose

This Document provides the second Annex to the main part of European ITS Framework Architecture Deliverable Document D 3.3, which provides a description of the European ITS Communication Architecture. This Annex Document includes additional and supporting information to the Main Document. It contains a description of the various technologies being currently used for ITS communications, an analysis of their main characteristics and a description of the OSI Model.

1.2 Where the document fits in the Architecture Documentation

The document is one of a set of two Annexes to the main European ITS Framework Architecture Communications Architecture Deliverable Document (D 3.3). The other Annex in the set is:

D3.3, Annex 1 - Communications Architecture - Supporting information for Communications Analysis

1.3 Document Structure

This Annex Document has been divided into seven Chapters, that follow this one. They provide information and data supporting the analysis of "example Systems" in the Main Document.

The following Chapter (2) provides a brief introduction to telecommunications technologies. It is followed by two Chapters (3 and 4), dealing with wireless technologies and wired technologies. There is then a Chapter (5) on optical communications technologies, after which there is a Chapter (6) containing a table summarising the main characteristics of all the technologies described in the three previous Chapters. Finally the last Chapter in this group (7) provides an overview description of the OSI Models. In all of these Chapters, the main emphasis is on the aspects of the technology that are relevant to ITS and the communications links that it requires.

1.4 Abbreviations

The following abbreviations have been used in this Document and their meanings may not be included in the adjacent text.

AuC	Authentication Centre
DSL	Digital Subscriber Line
DTE	Data Terminal Equipment
Gbps	Giga bits per second

GHz	Giga Hertz
IMEI	Internet Mobile Equipment Identity
ISL	Inter-Satellite Link
kbps	kilo bytes per second
Mbps	Mega bits per second
MFO	Multi-Function Outstation
MoU	Memorandum of Understanding
OSI	Open System Interconnection
TCH	Traffic Channel
TD-CDMA	Time Division – Code Division Multiple Access
TETRA	Terrestrial Trunked Radio
W-CDMA	Wideband - Code Division Multiple Access

2 Telecommunication Technologies

Telecommunications is one of the fastest developing technologies today. The main driving force for this is the permanent growing demand for more and better services by the customers. This was met by the government bodies with deregulation and the opening of national telecommunication markets. This lead to increasing competition with the result of new developments both in the used hardware and the used technologies.

As a result of this changes, ITS is faced with new telecommunication equipment and new telecommunication standards with an ever increasing speed.

The best example for this situation can be seen at the mobile handset market. Today's state of the art mobiles equipment will be outdated in one or two years. And each new family of mobile handsets shows changes in the interfaces, both in electrical and mechanical sense. Thus changes in ITS-applications (for example using GSM) must be done by each improvement of the product family of the mobile handset manufacturers.

The following chapters will offer a short abstract of the currently used telecommunication standards that are linked with ITS-applications and mention the possibly emerging standards of the near future.

3 Wireless Technologies

3.1 Introduction

In most cases, one important point in ITS communication is the fact, that often at least one user of an ITS system is on the move. The best (and usually only) way to communicate with the moving user is using wireless technologies.

Since radio spectrum is a limited resource shared by all standards and users, the wireless part of the communication within ITS systems can be seen as the most critical one in terms of bandwidth, security or costs.

Thus choosing the best fitted means of wireless communication may determine the overall success of any ITS application. And best fitted is not only restricted to considerations in a technical sense. Nowadays many marketing and commercial aspects must be taken into account before the technology decision can be done.

3.2 GSM

3.2.1 Introduction

GSM (**G**lobal **S**ystem for **M**obile Communication) is a digital personal mobile communication system based on a cellular network. Its main goal is to offer both the user and the operator the technical base for mobile speech and data communication.

During the development of this standard one main task was to secure interoperability and inter-working. By this the user should be enabled to use a service that today is called “International Roaming”. Looking at the active GSM networks this goal is reached for speech services, whereas the data services and several supplementary services still have many problems to solve.

3.2.2 Services provided by GSM

When GSM was initially planned, ISDN compatibility in terms of the services offered and the control signalling used was one of the main goals. However, radio transmission limitations, in terms of bandwidth and cost, made it too difficult to allow the ISDN B-channel bit rate of 64 kbps to be achieved practically. Today, this situation is about to change and technical improvements will rise that limitations up to 2 Mbps (see UMTS).

Next to the speech services GSM offers a variety of data services. GSM customers can send and receive data to and from users on POTS (Plain Old Telephone Service), ISDN, Packet Switched Public Data Networks, and Circuit Switched Public Data Networks.

Supplementary services are provided on top of teleservices or bearer services. At the moment, the following supplementary services are defined within the specification:

- CLIP / CLIR Calling Line Identification Presentation / Restriction
- COLP / COLR Connected Line identification Presentation / Restriction
- MPTY Multi Party
- CF Call Forwarding (different conditions possible)
- CW Call Waiting
- CUG Closed User Group
- AoC Advice of Charge
- CB Call Barring (different conditions possible)
- HOLD Call Hold

It must be noted that even if most of the European GSM networks have implemented many different supplementary services no customer can be assured that the supplementary services he is used to in his home network will work the same way in other networks. The reasons for that can vary from technical problems via commercial difficulties up to legal differences between these countries[2].

3.2.3 Multiple access and channel structure

The method chosen by GSM is a combination of Time- and Frequency-Division Multiple Access (TDMA/FDMA). The FDMA part involves the division by frequency of the total frequency band into several carrier frequencies. Each of these frequencies is spaced 200 kHz apart. Next to that different frequency bands are used for the uplink (communication leg from the mobile station to the base station) and for the downlink (communication leg from the base station to the mobile station). Due to the duplex communication of GSM there is always one uplink frequency dedicated to one downlink frequency.

The following frequencies are reserved for GSM within Europe:

- uplink: 890 – 915 MHz downlink: 935 – 960 MHz
- 1710 – 1785 MHz 1805 – 1880 MHz

In the USA a frequency band by 1900 MHz is defined. The use of other frequency bands (like 450 MHz) are currently being investigated.

One or more carrier frequencies, depending on the expected load within the range of the cell, are assigned to each base station. Each of these carrier frequencies is then divided in time, using a TDMA scheme.

The fundamental unit of time in this TDMA scheme is called a burst period and it lasts 15/26 ms (approximately 0,577 ms). Eight burst periods are grouped into one TDMA frame (120/26

[2] An example for legal differences can be seen in the rules for data protection. Whereas in Germany operators must pay attention to data protection regulations when implementing the supplementary service CLIP / CLIR comparable regulations do not exist in other European countries.

ms, approximately 4,615 ms), which forms the basic unit for the definition of logical channels. One physical channel is one burst period per TDMA frame.

Channels are defined by the number and position of their corresponding burst periods. All these definitions are cyclic. The entire pattern is repeated in a period of approximately 3 hours. Channels can be divided into dedicated channels, which are allocated to a mobile station, and common channels, which are used by mobile stations in idle mode.

3.2.4 Traffic channels

A traffic channel (TCH) is used to carry speech and data traffic. Traffic channels are defined using a 26-frame multi-frame, or group of 26 TDMA frames. The length of a 26-frame multi-frame is 120 ms, which is how the length of a burst period is defined (120 ms divided by 26 frames divided by 8 burst periods per frame). Out of the 26 frames, 24 are used for traffic, 1 is used for the Slow Associated Control Channel (SACCH) and 1 is currently unused. TCHs for the uplink and downlink are separated in time by 3 burst periods, so that the mobile station does not have to transmit and receive simultaneously. Using this method the electronic components of the mobile station can be simplified.

In addition to these full-rate TCHs, there are also half-rate TCHs defined[3]. Half-rate TCHs will effectively double the capacity of a system once they are implemented. For doing this specific half-rate speech coders must be implemented both within the GSM-network and the mobile stations using it. Next to the half rate coders enhanced full rate coders are defined. These coders offer a better speech quality to the users, thus this coders are implemented by several networks.

3.2.5 Control channels

Common channels can be accessed both by idle mode and dedicated mode mobiles. The common channels are used by idle mode mobiles to exchange the signalling information required to change to dedicated mode. Mobiles already in dedicated mode monitor the surrounding base stations for handover and other information. The common channels are defined within a 51-frame multiframe, so that dedicated mobiles using the 26-frame multi-frame TCH structure can still monitor control channels. The common channels include:

- Broadcast Control Channel (BCCH) - Continually broadcasts, on the down-link, information including base station identity, frequency allocations, and frequency-hopping sequences.
- Frequency Correction Channel (FCCH) and Synchronisation Channel (SCH) - Used to synchronise the mobile to the time slot structure of a cell by defining the boundaries of burst periods, and the time slot numbering. Every cell in a GSM network broadcasts exactly one FCCH and one SCH, which are by definition on time slot number 0 (within a TDMA frame).

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[3] They are not yet implemented within a live network.

- Random Access Channel (RACH) - Slotted Aloha channel used by the mobile to request access to the network.
- Paging Channel (PCH) - Used to alert the mobile station of an incoming call.
- Access Grant Channel (AGCH) - Used to allocate an SDCCH to a mobile for signalling (in order to obtain a dedicated channel), following a request on the RACH.

3.2.6 Authentication and security in GSM-networks

For the authentication procedure two functional entities of GSM-systems are involved, the SIM card located in the mobile, and the Authentication Centre (AuC) located in the network. Each subscriber is given a secret key. One copy of the key is stored in the SIM card and the other one is stored in the AuC. During the authentication procedure, the AuC generates a random number that it sends to the mobile. Both the mobile and the AuC then use this random number together with the subscriber's secret key and a special ciphering algorithm (called A3), to generate a signed response (SRES). This SRES is sent from the mobile station to the AuC. If this number is the same as the one calculated by the AuC, the subscriber is authenticated.

The same initial random number and the same subscriber key are also used to compute the ciphering key using an algorithm called A8. This ciphering key, together with the TDMA frame number, use the A5 algorithm to create a 114 bit sequence that is XORed with the 114 bits of a burst. Enciphering is an additional security option, since the signal is already coded, interleaved, and transmitted in a TDMA manner, thus providing protection from all but the most persistent and dedicated eavesdroppers.

Another level of security is performed on the mobile equipment itself, as opposed to the mobile subscriber. Each GSM mobile station is identified by a unique International Mobile Equipment Identity (IMEI) number. A list of IMEIs in the network is stored in the Equipment Identity Register (EIR). The status returned in response to an IMEI query to the EIR is one of the following:

- White-listed: The terminal is allowed to connect to the network.
- Grey-listed The terminal is under observation from the network for possible problems.
- Black-listed The terminal has either been reported stolen, or is not type approved (the correct type of terminal for a GSM network). The terminal is not allowed to connect to the network.

Thus the network provider is given a possibility to prevent the use of irregular (e.g. stolen or defective equipment) within its network. However, this method is not implemented by all network providers or some providers do not exchange their black list with other networks. In some countries this method is even used to block mobiles where the import duty is not paid from working within the local networks.

3.2.7 Speech Services

3.2.7.1 Speech coding

GSM is a digital system, so speech which is analogue, has to be digitised. The method employed by ISDN, and by current telephone systems for multiplexing voice lines over high speed trunks and optical fibre lines, is Pulse Coded Modulation (PCM). The output stream from PCM is 64 kbps. This rate is too high to be feasible over a GSM link. The 64 kbps signal, although simple to implement, still contains much redundancy.

Several speech coding algorithms were studied on the basis of subjective speech quality and complexity (according to cost, processing delay, and power consumption) before a Regular Pulse Excited -- Linear Predictive Coder (RPE--LPC) with a Long Term Predictor loop was chosen. Basically, information from previous samples, which does not change very quickly, is used to predict the current sample. The coefficients of the linear combination of the previous samples, plus an encoded form of the residual, the difference between the predicted and actual sample, represent the signal. Speech is divided into 20 millisecond samples, each of which is encoded as 260 bits, giving a total bit rate of 13 kbps. This is the so-called Full-Rate speech coding.

Enhanced Full-Rate (EFR) speech coding algorithm has been implemented by some European GSM1800 operators and North American GSM1900 operators. This offers an improved speech quality using the existing 13 kbps bit rate.

A Half Rate (HR) speech coding algorithm was defined by ETSI, too. This coding offers the possibility of effectively double the customers served within one cell by cutting in halve the needed bit rate. But this coding is said to show a poorer speech quality to the customer, thus the network providers have not implemented it yet.

3.2.7.2 Inter-working with supplementary services

Most of the supplementary services are applicable on the speech service of GSM.

3.2.7.3 Hand-over

When a call is in progress, the mobility of a GSM user may cause the need to switch the serving cell of the network especially if the quality of transmission drops below a certain value. Thus GSM defines a procedure called hand-over where a call (or any other connection) can be transferred from one cell to another. For this procedure quality parameters are defined so that this hand-over will not be noticed by the user or the disturbance will be kept to a minimum.

Hand-over procedures are defined for changes between cells that belong to the same and to different BSC / MSC regions of one network. Hand-overs between different networks (e.g. between two different countries) are not possible yet.

3.2.8 Data (circuit switched data)

The GSM standard supports different types of circuit switched data services. These bearer services can be transparent and non transparent, synchronous and asynchronous. This duplex service usually offers a bit rate between 300 and 9600 bit / s. During the usage of the service the achievable bit rate depends on the quality of the radio link, thus in many cases the bit rate will be below the achievable maximum. Some networks even have implemented a data service that offer a bit rate up to 14400 bit / s, but the result is an increased bit error rate in non ideal communication situations.

3.2.9 SMS - Point to Point

A special feature of GSM is the Short Message Service (SMS). These feature can be summarised:

- SMS is a bi-directional service for short alphanumeric messages (up to 160 characters using a special 7-bit alphabet) or transparent data (up to 140 byte per message).
- Short messages are transported in a store-and-forward fashion, they are not sent directly from the sender to the recipient. The messages are sent via a special entity within the GSM-network, the so called SMSC (Short Message Service Centre). Thus if the recipient has switched off the mobile the message becomes available to him as soon as the mobile station is switched on again.
- The notification of the reception of a messages can be send by the SMSC if this is asked for by the sender.
- SMS can be sent and received simultaneously to GSM voice, Data and Fax transmission . This is possible because SMS are transferred via signalling channels. Thus SMS can be used when both voice and data communication is needed to perform a service.
- Typically short messages are stored within the GSM SIM-card. Most mobile phones offer a serial interface. Usually one of the features of this interface is the transfer of short messages to terminal equipment. Special parameters can define the SMS to be transferred directly to this terminal equipment.
- The transfer of short messages between different GSM-networks (roaming or recipient in an other network) is defined within the GSM-standard. However, this possibility is sometimes blocked by some network provider.
- For larger customers (e.g. service centres in the sense of ITS) direct access to SMSCs are supported. However, in most cases different access protocols are used by the SMSC manufacturer.

For point-to-point SMS, a message can be sent to another subscriber to the service, and an acknowledgement of receipt can be provided to the sender.

3.2.10 SMS - Cell Broadcast

The Short Message Service can also be used in a cell-broadcast mode. In this mode only mobile terminated message up to a length of 82 bytes can be sent. Thus this service is perfectly fitted for sending short messages such as traffic updates or news updates to a huge amount of recipients.

Two different kinds of cell broadcast information can be defined: static and dynamic information. Implementing static CB information within the networks can be done within the BTS itself. For dynamic CB information a Cell Broadcast Centre (CBC) is needed where the permanent change of transmitted information is managed.

ETSI defined this service only to be optional within the GSM networks. Thus only some networks have implemented this service. In addition many older mobile handsets may not be able decode and display the CB information.

3.2.11 GPRS

GPRS (General Packet Radio Service) is an upgrade to the GSM standard and is widely seen as a first step towards a 3rd generation mobile standard.

Because GPRS is based on a packet switched protocol, data is sent in small packets, this service has similarities witch the Short Message Service. Packet Switching means that the radio resources are only used when the user is sending or receiving data. This enables connectionless applications where the user does not always transmit data. But the application is ready to do so immediately when the need arises. Using this method the radio resources are used more efficiently, so many GPRS users can share the same bandwidth and can be served by a single cell.

GPRS will be able to allow data transmission speeds up to 115 kbps (in theory). This depends on the used data coding scheme, the amount of bundled GSM timeslots and the amount of active GPRS users within the serving cell.

3.2.12 HSCSD

HSCSD (High Speed Circuit Switched Data) is a technology that bundles several timeslots (channels) of one cell and uses them for the data communication of one customer. Thus the HSCSD-user can increase its maximum usable bandwidth according to the amount of timeslots used. Theoretically there can be eight channels assigned to one connection. However due to the existing network architecture the current maximum is four. Thus the network provider can offer a bandwidth of up to 38,4 kbps (on a base of 9,6 kbps per channel).

Next to the bundling of timeslots some network providers are implementing a new channel coding. As an advantage this coding increases the usable bandwidth per channel from 9,6 kbps to 14,4 kbps. Thus these networks will offer a total bandwidth of 56 kbps to their HSCSD customers. On the other hand the new coding reduces the robustness of the transmission. The result is a higher bit error rate if the quality of radio reception is poor.

One problem occurs when the user in a multi-timeslot environment performs a handover (dynamic call transfer between neighbouring cells). In such situation the bandwidth may drop due to lower number of unused timeslots in the new cell during the handover.

From the commercial point of view a direct trade-off between increased speed and the costs from using more radio resources can be seen. However at least one network provider announced tariffs with no difference between HSCSD and normal circuit switched data.

Currently only a few GSM-networks are implementing this technology, most networks and handset manufacturers prefer GPRS. Thus the possible use of HSCSD for ITS may be seen in the market development of the future.

3.2.13 GSM-R (PMR-System)

GSM-R (GSM-Railway) is a special improvement of the GSM-Standard to fulfil the requirements of PMR-services for railways.

The standard GSM services are added with special Features (usually named "Advanced Speech Call Items"). Some of these Advanced Speech Call Items are:

- Group Calls
- Emergency Calls with priority handling and pre-emption
- Addressing via train numbers (special service for railways)
- very high network availability (> 99,9 %)
- short call establishment (< 5s)
- low call cut ratio
- support of speeds up to 500 km/h

Using these features special applications can be implemented that fulfil the needs of railway organisations. Some examples for these applications are train security and safety, train control systems, facility management or special travellers information services.

In Germany the Deutsche Bundesbahn is implementing a GSM-R system. This system is using 19 duplex channels at the frequencies between 876 to 880 MHz (uplink) and 921 to 925 MHz (downlink). Until the end of 2002 all existing analogue PMR-systems used by the Deutsche Bundesbahn will be replaced by GSM-R.

The main disadvantages of GSM-R are:

- The system cannot be used by the public GSM-users. Thus public GSM users still must use the public networks when their telephone is used within a train.

- GSM-R does not support direct mode. Direct mode is a feature offered by the new digital PMR-systems like TETRA and Tetrapol where mobiles can communicate with each other without the support of any network infrastructure. This feature is very important for emergency organisations like police, ambulance and fire brigade.

3.2.14 EDGE

EDGE (Enhanced Data-rates for GSM Evolution) is a high speed mobile data standard which is enhancing channel coding for GSM-Systems (e.g. GPRS) in order to achieve data transmission speeds of up to 384 kbps.

For EDGE only small changes in the current technology of the mobile networks are needed. By the use of an enhanced channel coding the capacity of one single timeslot is increased to a maximum of 48 kbps. In addition several timeslot can be used by one mobile at the same time increasing the total bandwidth to a theoretical maximum of 384 kbps.

EDGE was developed to meet the requirements of mobile network operators who fail to win UMTS spectrum.

3.3 UMTS

UMTS (Universal Mobile Telephone System) is a part of the International Telecommunications Union's 'IMT-2000' vision of a global family of 3rd generation mobile communications systems. UMTS will play a key role in creating the future mass market for high-quality wireless multimedia communications that should be offered World-wide by the year 2010.

It will provide an evolution in mobile data with data transmission rates up to 2 Mbps and will deliver low-cost, high-capacity mobile communications with global roaming and other advanced capabilities

UMTS experimental systems are now in field trial. It is expected that UMTS services will be launch commercially from 2002 on.

UMTS will deliver pictures, graphics, video communications and other wide-band information as well as voice and data, direct to people who can be on the move. UMTS will build on and extend the capability of today's mobile technologies (like digital cellular and cordless) by providing increased capacity, data capability and a far greater range of services using an innovative radio access scheme and an enhanced, evolving core network.

3.4 W-CDMA and TD-CDMA

W-CDMA (Wideband - Code Division Multiple Access) and TD-CDMA (Time Division - Code Division Multiple Access) are technologies that share the common goal of allowing the maximum number of calls to simultaneously take place:

- TDMA (Time Division Multiple Access) is a method of digital wireless transmission, allowing many users to access (in sequence) a single radio frequency channel without interference by allocating unique time slots to each user within each channel. With TDMA, each user has access to the entire radio frequency channel for a brief period, during which they can transmit data. This frequency channel is shared with other users who have time slots allocated at different times. TDMA enables service providers to increase the number of calls processed without increasing the quantity of supporting network equipment.
- CDMA (Division Multiple Access) is a spread spectrum technique for multiple access. In CDMA, each user is assigned a pseudo-noise (PN) code to modulate transmitted data. The PN code is a long sequence of ones and zeros similar to the output of a random number generator of a computer: although the numbers are not really random--they are generated using a specific algorithm--they appear to be

random. Because the codes are nearly random, there is very little correlation between the different codes. In addition, there is very little correlation between a specific code and any time shift of that same code. Thus, the distinct codes can be transmitted over the same time and the same frequencies and the signals can be decoded at the receiver by correlating the received signal (which is the sum of all transmitted signals) with each PN code.

- W-CDMA uses a broader frequency spectrum channel (of 5MHz, and potentially up to 20MHz) that acts as a high-capacity wide-band channel to enable higher data-rate transmissions for mobile communications.

3.5 DECT

3.5.1 Introduction

DECT (Digital Enhanced Cordless Telecommunication) is a common standard for cordless personal telephony established by ETSI (European Telecommunications Standards Institute). Its micro cellular concept basically aims on the indoor use of mobiles in an environment with a high subscriber density, approximately up to 10000 Erlang (“talkers”) per square kilometre.

The maximum distance between base- and mobile station is estimated to be 50 meters indoors and up to 300 meters outdoor. The maximum movement speed is between 20 and 50 km / h.

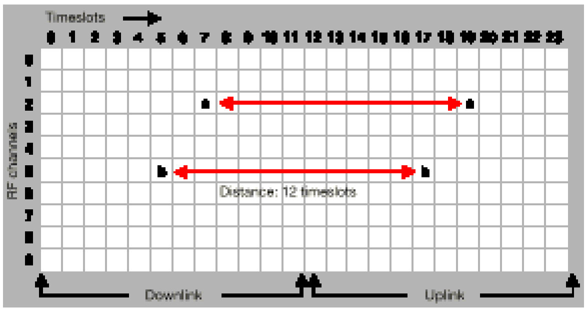
The main benefits of DECT are:

- Robust self organising real time radio channel selection
- Coexistence
- Multiple access rights
- Seamless hand-over
- Mobility
- High subscriber densities
- Inter operability
- High quality voice
- Cost efficient
- High level of security
- Standard inter networking profiles
- Multimedia terminals
- Flexible bandwidth

3.5.2 Description

A DECT system comprises a fixed part, utilising one or more base stations, and one or more portable parts. In first way the DECT standard only defines the air interface between a fixed part and a portable. The radio interfaces uses a Multi Carrier, Time Division Multiple Access, Time Division Duplex (MC/TDMA/TDD) access method with a continuous Dynamic Channel Selection and Allocation capability. This enables a high capacity for a pico-cellular systems. DECT can even be utilised in busy or noisy radio environments. These methods enable DECT to be used without the need of expensive frequency planning. DECT makes efficient use of the assigned radio spectrum, even when multiple operators and applications share the same frequency spectrum.

Figure 1 DECT transmission – frequency allocation and time spectrum



Basic DECT frequency allocation uses 10 carrier frequencies in the 1880 to 1900 MHz range. The time spectrum for DECT is divided into time frames repeating every 10 ms. Each frame consists of 24 time slots each individually accessible that may be used for either transmission or reception. For the basic DECT speech service two time slots - with 5 ms separation - are paired to provide a full duplex bearer capacity of up to 32 kbps. For easy implementation the 10 ms time frame is split in halves. The first 12 time slots are used for the downlink, the other 12 are used for the uplink. This structure allows up to 12 simultaneous basic DECT (full duplex) voice connections per transceiver.

DECT is able to combine several channels into one single bearer. For data transmission purposes a throughput rate of n x 24 kbps (with error protection) can be achieved, up to a maximum of 552 kbps with full security.

3.5.3 Dynamic Channel Selection and Allocation

All DECT equipment is recommended to scan its local radio environment at least every 30 seconds. This means that the local RF signal strength on all idle channels is measured. Scanning is done as a background process and produces a list of free and occupied channels to be used in the channel selection process. With the aid of the information of this list a DECT station is able to select the best fitted (least interfered) channel to set up a new communication link. In a portable part, the channels with the highest signal strength are continuously analysed to check if the transmission originates from a base station to which the portable has access rights. The portable will lock onto the strongest base station. Channels with lowest signal strength are used to set up a radio link with the base station if the portable user decides to establish a communication or when an incoming call is signalled to the portable. Thus the communication links are established on the channel with lowest actual interference.

3.5.4 Hand over

DECT portables can escape from an interfered radio connection by establishing a second radio link on a newly selected channel to either the same or to another base. Both radio links are temporarily maintained in parallel with identical transmissions. The quality of these links is analysed. Then the base station finds out the radio link with the best quality. The other one is released.

If the portable is moving from one cell area into another, the received signal strength of the two base stations are changing. When the new base stations signal becomes stronger than the signal from the old base station, a seamless hand over (see above) will be performed. The seamless hand over is a fully autonomous initiative from the portable. The user will not notice this procedure..

Although a hand over is always initiated by the portable, it may also be the uplink that suffers from poor quality. For this case, the signalling protocol enables the base station to signal the link quality to the portable. Then the portable can initiate the hand over procedure.

3.5.5 Security

The use of a radio access technology providing mobility includes considerable risks with respect to the security of the system. The DECT standard provides several measures to counteract these technological flaws in security. Special subscription and authentication protocols have been included to prevent unauthorised access to the DECT system. In addition a ciphering concept can be used that will provide protection against many methods for eavesdropping.

3.5.6 Subscription

The subscription procedure is the procedure where the DECT network opens the service to a portable.

The network operator provides the user of any portable with a PIN code. This PIN will be entered into the fixed and the portable part before the procedure starts. Next to that the portable must know the identity of the fixed part it will subscribe to.

This subscription procedure can be done "over the air": A radio link is established and both ends verify that they use the same subscription key. Portable and network identities are exchanged. Then both sides calculate the secret authentication key that should be used for authentication at each single call set up. This authentication key is only known to the portable and the fixed part, it is not transferred over the air.

A portable may have several possible subscriptions with different authentication keys for each DECT network. All keys and network identities are stored in a list within the portable. Any portable will only communicate within a network where it has valid access rights.

3.5.7 Authentication

Authentication of a handset may be done as a standard procedure at each call set up. During the authentication procedure, the base station checks the secret authentication key by a challenge-response-procedure:

The base station sends a random number to the handset. The handset calculates a response by using the authentication key and the challenge number as input parameters. This response is transmitted to the base station. The base station also calculates the expected response and compares it with the received response. If successful the call set up is continued, otherwise the call attempt is rejected.

3.5.8 Encryption

During authentication, both sides also calculate a cipher key. This key can be used to cipher the data sent over the air. At the receiving side the same key is used to decipher the information. The ciphering process is an optional part of the standard.

3.5.9 Profiles

Application profiles contain additional specifications defining how the DECT air interface should be used in specific applications. The most important profiles are:

- GAP: Generic Access Profile, minimum mandatory set of technical requirements
- GIP: GSM / DECT Inter-working Profile
- IAP / IIP: ISDN Inter-working Profile
- RAP: Radio local loop Access Profile, profile for the “last mile”
- CAP: Cordless terminal mobility Access Profile, profile for the roaming of DECT portables.

3.6 TETRA (PMR-System)

3.6.1 Introduction

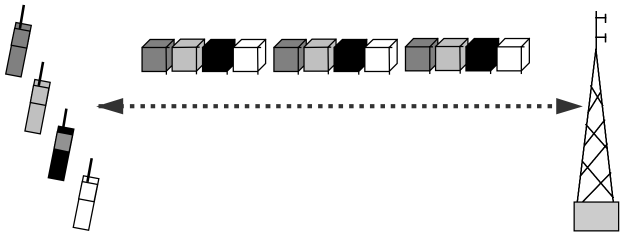
TErrestrial **T**runked **RA**dio (TETRA) is an open digital trunked radio standard which is defined by the European Telecommunications Standardisation Institute (ETSI) to meet the needs of most professional mobile radio users.

The TETRA Memorandum of Understanding (MoU) was founded in 1994. It represents a forum for all interested parties which are professional users, manufacturers, network operators, test houses and telecom agencies.

3.6.2 Description

TETRA supports voice, circuit switched data and packet switched data services with a different data transmission rates and several levels of error protection. TETRA uses TDMA (Time Division Multiple Access) technology with four user channels interleaved into one carrier with 25 kHz carrier spacing. Higher data transfer rates up to 28,8 kbps are implemented by reserving up to four channels for the same user connection. The needed bandwidth is allocated on demand.

Figure 2 TETRA Illustration of Channel Organisation



TETRA is designed as a trunked system that effectively and economically supports shared usage of the network by several organisations. However privacy and mutual security is supported by this system. Virtual networking inside the TETRA network enables each organisation to operate independently, but still enjoy the benefits of a large, high-functionality system with efficient resource employment.

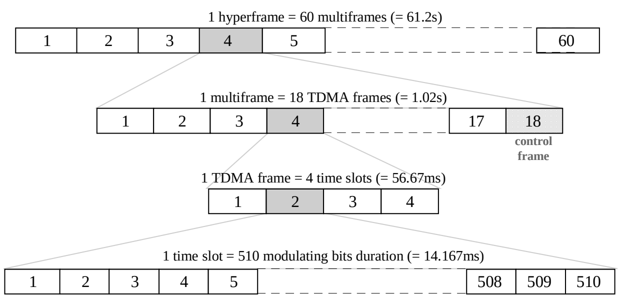
TETRA is a high security technology that inherently includes encryption of voice, data, signalling and user identities. Two encryption mechanisms are defined:

- air interface encryption; the radio path between the terminal and the base station is encrypted by the network provider.
- end-to-end encryption; the most critical applications can use encryption methods for the transmission throughout the system to the other terminal.

TETRA provides a fast call set-up time (approximately 300 ms). This feature is crucial especially for the public safety and emergency services. TETRA supports both semi-duplex operation for efficient group communication and duplex operation for telephony type individual calls. The feature “Advanced Group Call” and “Announcement Call” included in TETRA meet the needs of the most critical user applications. “Multiple Call Priority Schemes” ensure effective resource allocation to very urgent traffic in the network that may be generated by emergency services.

The TETRA frame structure has four time slots per TDMA frame. This is further organised as 18 TDMA frames per multi-frame. In circuit mode both voice and data traffic from approximately 1 second (= 1 multi-frame length of time) is compressed and transmitted in 17 TDMA frames. Thus the 18th frame can be used for control signalling without interrupting the flow of user data. This 18th frame is called the control frame and provides the basis for slow associated control channel (SACCH). The SACCH provides the background control channel signalling.

Figure 3 TETRA Frame Structure



The gross bit rate of one channel is 9 kbps, into which speech is coded with 4,8 kbps net bit rate using ACELP coding. The modulation method used in TETRA is π/4-DQPSK (Differential Quaternary Phase Shift Keying).

Within the TETRA standard direct mode operation between mobile radios without the need for network infrastructure is defined, too. Also repeater and gateway functions are defined. These features will extend the coverage of hand portable radios in both direct mode and network operation.

The defined power classes of TETRA radio equipment are 25 W, 10 W, 3W and 1 W. TETRA radios can automatically adjust the output power according to the needed field strength. Thus energy consumption is reduced and operation time is increased.

Several teleservices provide complete communication capability for between users, including all terminal functions. In TETRA standards teleservices cover voice communications services.

3.6.3 TETRA Teleservices

- Individual Call
- Group Call
- Acknowledged Group Call
- Broadcast Call

A bearer service provides communication capability between terminal network interfaces, excluding the functions of the terminal. TETRA bearer services are defined for data transfer.

3.6.4 TETRA Bearer Services

- Circuit mode data 7,2/14,4/21,6/28,8 kbps
- Circuit mode protected data 4,8/9,6/14,4/19,2 kbps
- Circuit mode heavily protected data 2,4/4,8/7,2/9,6 kbps
- Connection oriented packet data
- Connectionless packet data

In addition, the standard defines supplementary services for very flexible system applications. Supplementary services modify or supplement the above mentioned services. Some supplementary services are essential, others are optional:

3.6.5 Essential Supplementary Services

- Access Priority: The uplink access of a radio unit can be handled with a higher priority when the serving network is congested.
- Priority Call: The access to the network resources can be prioritised, too.

- Pre-emptive Priority Call: This call has the highest uplink priority and highest priority access to network resources. If the system is busy the communication with the lowest priority will be dropped to allow this call to continue.
- Call Authorised by Dispatcher: A dispatcher verifies the validity of a call request before the call is allowed to proceed.
- Area Selection: Areas of operation within the network are defined for the user. It can be redefined on a call by call basis.
- Late Entry: Latecomers may join a call in progress
- Discrete Listening: Authorised radio unit may monitor a communication without being identified
- Ambience Listening: Dispatcher may turn on the transmitter of a radio unit without any indication being provided on radio unit.
- Dynamic Group Number Assignment: The dispatcher is able to program new group numbers into the radio units over the air interface.

3.6.6 Optional Supplementary Services

- Calling Line Identification Present: The unit ID of calling party is presented.
- Connected Line Identification Present: The unit ID of the called party is presented.
- Calling/Connected Line Identification Restriction: Either party may prevent the presentation of its unit ID.
- Call Report: Displays calling party ID on a busy radio unit.
- Talking Party Identification: radio unit automatically identifies itself when it is in a group call.
- Call Forward Unconditional: Allows a radio unit to forward all calls to another radio unit.
- Call Forward on Subscriber Busy: Allows a radio unit to forward calls if the radio unit busy.
- Call Forward on Subscriber Not Reachable: Allows a radio unit to forward calls when it is out of service or switched off.
- Call Forward on No Reply: Allows a radio unit to forward all unanswered calls.
- List Search Call: Incoming call will sequence through a user defined list until the call is answered.
- Short Number Addressing: Short number dialling
- Call Waiting: Notification of an incoming call to radio unit which is busy.
- Call Hold: Allows user to interrupt an existing call and re-establish the call when required.
- Call Completion Busy Subscriber: An incoming call will wait until subscriber is free before calling back.

- Call Completion No Reply: An incoming call will wait until the subscriber has made a call before calling back
- Include Call: Ability to include a radio unit in an existing call
- Advice of charge: Call charge information at start, during or end of call
- Call Barring: Ability to bar a call from/to a user defined list
- Call Retention: Ability to prevent call being pre-empted
- Transfer of Control: Initiator of a group call can transfer ownership to another party

3.7 TETRAPOL (PMR-System)

TETRAPOL is a digital trunked radio system comparable to TETRA above. Nevertheless TETRAPOL's primary target market were the public safety organisations

3.8 Satellite Communications Systems

3.8.1 Introduction

Satellite Communication Systems offer a family of services that enables the users to make personal (mobile) communications all over the world (or in huge areas) with the same communication system. The availability of current ground-based cellular communications is limited, for geographical and technical reasons. New systems like Iridium, Globalstar and ICO have begun to complement these networks and by extending their coverage across the globe

3.8.2 Iridium

(Note: Iridium Corporation has filed for Chapter 11 protection due to financial debts.)

Iridium is a satellite based personal communications system. Founded by Motorola it is designed to allow telecommunication service from any place on earth, thus even the polar regions are covered. At the beginning Iridium was planned to consist of 77 satellites[4], later the system was redesigned to use only 66 satellites. Iridium is a so called LEO-system (Low Earth Orbiting), where the satellites orbit at 780 km altitude in circular orbits. LEO-satellites were chosen because they offer both low path delays and global coverage.

Iridium uses Inter-Satellite Links (ISL), a telephony architecture based on GSM and a geographically-controlled system access process. Each satellite can utilise up to 48 beams ('footprints'), making a total maximum of 3168 beams for the Iridium system. Out of these only approximately 2150 will be active at one time, because some beams are switched off when the satellite crosses the polar region (here beam overlap may occur).

[4] 77 is the atomic number of the element iridium. This way the name for this system was found.

Connections between the Iridium system and the public switched telephone network are provided via gateway installations. Each satellite is connected to its four neighbouring satellites by inter-satellite links (ISL). These ISLs provide flexibility in where the gateways can be located. Thus an Iridium call can be routed within the satellite network and connected to any mobile located anywhere, or it can be connected to the public network through any gateway.

The Iridium system patterns its call processing architecture after GSM, and the gateways incorporate a GSM MSC with the associated databases (EIR, HLR, VLR). Additional functions required for the Iridium system and not accommodated by the GSM MSC are also taken care of in the gateways.

When a mobile originates a call, the Iridium system will calculate the user's location. Each gateway is associated with a location area which the gateway controls, and the mobile location is used to assign the home (or visited) gateway which controls all aspects the call. If required, a PSTN/PLMN connecting gateway is chosen based upon the mobile's location and the location of the PSTN/PLMN party at the time of call set-up. Using the position of the mobile, the gateway will ensure the compliancy with national laws enforcing call restrictions on mobiles.

The Iridium satellite system provides multiple services including telephone transmission voice with 2,4 kbps, data with 2,4 kbps, fax, or paging.

The following frequencies are assigned to the Iridium system:

user frequency:	1,616 GHz to 1,6265 GHz.
feeder uplink frequency:	29,1 GHz to 29,3 GHz.
feeder downlink frequency:	19,4 GHz to 19,6 GHz.
Inter-satellite cross-links frequency:	23,18 GHz to 23,38 GHz.

3.8.3 Globalstar

3.8.3.1 Introduction

Globalstar, organised as a consortium of leading international telecommunications companies, is starting a low-earth-orbiting (LEO) satellite digital telecommunications system end of 1999. This system offers wireless telephone and other telecommunications services World-wide delivered by satellites. The Globalstar mobile stations are designed to be multi-mode.[5]

The services offered to the customers by Globalstar include:

- Voice calling
- Short Messaging Service (SMS)

[5] Multi-mode mobile stations are equipment which can be used for different communication technologies. In this section this means that a handset can be used either for satellite services or for terrestrial cellular service, including GSM, AMPS, and CDMA. The use depends on the actual availability of the service.

- Roaming
- Positioning (planned for 2000)
- Facsimile (planned for 2000)
- Data transmission (planned for 2000)

3.8.3.2 Technology

The Globalstar system is designed to provide satellite-based telephony services to a broad range of users. Globalstar meets the needs of cellular users who roam outside of coverage areas, people who work in remote areas where terrestrial systems do not exist, residents of under-served markets who can use Globalstar's fixed-site phones to satisfy their needs for basic telephony, and international travellers who need to keep in constant touch.

Globalstar's phones look and act like mobile or fixed phones. The difference is that they can operate virtually anywhere, carrying the call over an Code Divisional Multiple Access (CDMA) satellite signal.

The 48 Low Earth Orbiting (LEO) satellites picks up signals from over 80% of the Earth's surface, everywhere outside the polar regions and some mid-ocean regions. Several satellites pick up a call simultaneously, and this "path diversity" assures that the call does not get dropped even if a phone moves out of sight of one of the satellites. If buildings or terrain block the signal, a handover takes place, and the call's transmission is switched to an alternative satellite with no interruption. This satellite now maintains transmission of the signal to one of several terrestrial gateways.

The Gateways process calls, then distribute them to fixed and cellular networks.

3.8.3.3 Satellite Constellations

The Globalstar constellation will expand to 48 satellites by the end of 1999, plus an additional four spare satellites will be in orbit.

The satellites are placed in eight orbital planes of six satellites each, inclined at 52 degrees to provide service on Earth from 70 degrees North latitude to 70 degrees South.

3.8.3.4 Gateways

Gateways are an integral part of the Globalstar ground segment, which also includes Ground Operations Control Centres (GOCCs), Satellite Operations Control Centres (SOCCs), and the Globalstar Data Network.

A gateway may service more than one country. Gateways consist of three or four dish antennas, a switching station and remote operating controls. Because all of the switches and complex hardware are located on the ground, it is easier for Globalstar to maintain and upgrade its system, more than it is for systems which handle switching in orbit (like Iridium).

Gateways offer seamless integration with local and regional telephony and wireless networks. Encryption ensures voice and signalling security for individual transmissions.

3.8.3.5 Globalstar Advantages

Low Voice Delay: The use of Low Earth Orbiting (LEO) satellites, moving at an altitude of 1,414 kilometres, means the customer is almost not able to make out the voice delay.

Reliability: The Globalstar system's software resides on and is monitored from the ground, not from satellites, which means easy and fast system maintenance and upgrades.

Fewer Dropped Calls: Globalstar's smart system design is based on overlapping satellite coverage, so that several satellites can be available from any location to handle a call. This "path diversity" results in a greater likelihood of establishing calls for customers and far less chance of dropping calls than other systems.

Lighter, All-in-one Handsets: Mobile phones manufactured by Ericsson, Qualcomm, and Telit are lightweight and can be used for either cellular or satellite-based calls. The phones are available with many accessories, such as car and marine kits, which extend the phone's reach still further.

3.8.4 ICO (Inmarsat)

(Note: ICO corporation has filed for Chapter 11 protection due to financial debts. ICO stocks were suspended trading in August and finally de-listed from NASDAQ in December 1999 Thus the following section may become obsolete very soon.)

3.8.4.1 Overview

Starting in first quarter 2001, ICO will introduce its satellite services. A fully integrated space/ground system will deliver digital voice, data, fax and messaging to and from users all over the world through handheld ICO phones similar in appearance, size and weight to today's GSM phones. The company also aims to serve rural and remote communities, aircraft, ships, land transport, public services and governments.

3.8.4.2 ICO's markets

ICO satellite system is designed to offer service in four different market segments:

- Cellular services: Existing users of terrestrial cellular networks who want to extend the service area within which they can roam. Carrying dual-mode ICO phones, these users will roam from terrestrial coverage to the ICO network and will be billed for their use of the satellite system by their cellular network operators.
- Basic mobile: People who need personal mobile communications but live or work outside cellular coverage areas. This service will be distributed by ICO service partners and other local distributors.
- Speciality mobile: Commercial and public service operators with a requirement for mobile communications aboard small craft, ships, medium/long-haul aircraft and land vehicles. Service will be delivered to these users who are expected to include governments and the energy industries via specialised non-handheld terminals.

- Semi-fixed: Terminals are being developed to meet the fixed communications needs of commercial operations and communities lying beyond the reach of terrestrial wired or wireless networks. Terminals will either take the form of stand alone desktop or payphones or will act as local wireless hubs, offering access to national and international networks.

3.8.4.3 ICO phones and services

Most ICO phones are expected to be dual-mode, working with both the satellite system and with terrestrial cellular networks. Users will be able to select a preferred terrestrial cellular standard (GSM, D-AMPS, PDC) at time of purchase. The basic ICO phone will be a handheld similar in size, weight and design to current pocket-sized cellular units.

ICO will offer digital voice, data, fax and messaging services, beginning in first quarter 2001.

Some terminal types are expected to include commercial vehicle, maritime, aeronautical, semi-fixed and supervisory control and data acquisition (SCADA) units.

3.8.4.4 System

ICO's system will consist of a space segment and a dedicated ground network. Called ICONET. The latter will provide the link between the satellites and existing public fixed and mobile networks.

The ICO space segment will operate with ten satellites and additional two in-orbit spare satellites. The system will operate in middle earth orbit (MEO) at an altitude of 10,390 km. Divided equally between two orthogonal planes, each inclined at 45 degrees to the equator, they will provide complete, continuous overlapping coverage of the Earth.

The satellites will operate in S-band and C-band, using digital onboard processing and time-division multiple access (TDMA). Each satellite will be able to handle up to 4,500 simultaneous phone calls.

The satellites will communicate with ground networks through the ICONET. This will consist of 12 Earth stations or satellite access nodes (SANs) located around the globe, and the high-capacity terrestrial links between them.

The SANs will provide the primary interface with the ICO satellites for routing traffic. The SANs will also link with points of interconnection that will serve as the primary interface with public switched telephone, mobile and data networks.

3.8.4.5 ICO's advantages

A higher satellite orbit will mean less likelihood of signal blockage and fewer call hand-offs, and thus superior quality of service. A smaller constellation will combine with longer satellite lifetimes to reduce the cost of space-segment ownership. The system works without any complex inter-satellite link.

With a minimum of hubs and concentrating intelligence in the ground network rather than the satellites, ICO will fully exploit the technical capability, versatility and economics of existing terrestrial fixed and mobile networks.

ICO is owned by telecommunications services companies which already have strong existing customer bases. No one nation or nationally based investor has a dominant interest. And the company's infrastructure is being procured not from a single manufacturer with proprietary technology but from a team of world-class industrial partners.

ICO will operate through strong local distribution partners backed up by globally dispersed business operations. Rather than creating its own, it will exploit existing distribution channels to market its products and services.

3.8.5 Orbcom

Orbcom is a satellite communication system which only offers data services to the customers. Theses services are messaging, email, fax and GPS. The commercial service began on 30 November 1998. In total 48 LEO satellites are used by the system. They are orbiting at 825km altitude and are offering a world-wide coverage. Eight additional satellites are kept as spares on the ground and may be added if there is sufficient demand.

The lifetime of each satellite is estimated to four years. The receiving frequency is between 148 MHz and 149,9 MHz, the transmitting frequency are both between137MHz and 138MHz and between 400,05 MHz and 400,15 MHz.

The used data rate is up to 2,4 kbps (typically 0,3 kbps).

3.9 Satellite Positioning Systems

3.9.1 Introduction

For many ITS services it is quite important that a most accurate positioning of the user can take place. The best examples for this are emergency and breakdown services where an exact position is needed. This can be provided specialised positioning systems.

Global Positioning Systems (GPS) are space-based radio positioning systems that provide 24 hour three-dimensional position, velocity and time information to suitably equipped users anywhere on or near the surface of the Earth (and sometimes off the earth). Global Navigation Satellite Systems (GNSS) are extended GPS systems, providing users with sufficient accuracy and integrity information to be useable for critical navigation applications. The NAVSTAR system, operated by the U.S. Department of Defence, is the first GPS system widely available to civilian users. The Russian GPS system, GLONASS, is similar in operation and may prove complimentary to the NAVSTAR system.

3.9.2 Satellite Navigation

Satellite navigation is a tool to determine position, velocity and precise time world-wide. Signals transmitted from several satellites are picked up by a receiver and used to calculate position, velocity and time. User navigation receivers measure the distance of the receiver to the satellite using a technique called 'passive ranging'. The distance to each satellite is derived from the measurement of the time taken for the navigation signal to travel from the satellite to the receiver. A three dimensional position can be calculated from available signals from three satellites, although a fourth is also required to eliminate the need for a precise atomic clock at the user.

3.9.3 GPS

GPS stands for Global Positioning System. Users of a global positioning system can determine their location anywhere on the earth. There are two "public" GPS systems. The NAVSTAR system (USA) and the GLONASS system (Russia).The NAVSTAR system is often referred to as ‘the GPS’ because it was the first public available system. Nevertheless, both systems are GPS systems.

Both systems provide two sets of positioning signals. The higher accuracy signal is reserved for each country's military use. The lower accuracy signal is available to civilian users free of charge.

A global positioning system uses the characteristics of radio transmissions for localisation. Satellite based transmitters are used to cover earth with signals needed for a high accuracy localisation. The information the satellites transmit are actual system time, satellite location and satellite status. A special radio receiver - a GPS receiver - to receive the transmissions from the satellite is needed. The GPS receiver contains a special computer that calculates the location of the receiver based on the information from the satellites.

The NAVSTAR system consists of 24 active satellites. Four satellites are located in each of six orbits. The orbits are distributed evenly around the earth, and are inclined 55 degrees from the equator. Each satellite transmits two different signals: L1 (1575.42 MHz) and L2 (1227.60 MHz). L1 is modulated with two pseudo-random noise signals - the protected (P) code, and the coarse/acquisition (C/A) code. L2 carries only the P code. Each satellite transmits a unique code for identifying the signals. When a feature called "Anti-Spoofing" is active, the P code is encrypted. Civilian receivers only use the C/A code on the L1 frequency.

The receiver measures the time required for the signal to travel from the satellite to the receiver. This is calculated by the time that the signal left the satellite and the time it is received. If the receiver had a perfect clock, synchronised with those on the satellites, three measurements from three different satellites would be sufficient to calculate the location in 3 dimensions. As you can't get a perfect clock that will fit in a low cost GPS receiver a fourth satellite is needed to correct the error caused by the receiver clock. If more than these four satellites can be received the calculation can be done with a higher accuracy.

Each measurement ("pseudo range") gives a position on the surface of a sphere centred on the corresponding satellite. Due to the receiver clock error, the four spheres will not intersect at an exact point. Thus the receiver must adjust its clock until the correct intersection point is found.

The Standard Positioning Service (SPS) which is available to civilian users should give 20 metre horizontal accuracy. However this accuracy is normally reduced to 100 metres (95% of the time) by Selective Availability (SA). The vertical accuracy is about 1.5 times worse than horizontal.

Selective Availability is an intentional degradation of accuracy. It is intended to prevent "the enemy" from making tactical use of the full accuracy of GPS. SA is normally switched on. Military receivers can use the encrypted P code to get 20 metre accuracy, or better, regardless of the state of SA.

Since the receiver must adjust its clock to be precisely synchronised with GPS time, a GPS receiver can be used as an exact time reference. The GPS system time (as used by the system itself) does not include leap seconds. Thus the difference between GPS time and UTC is included in the information sent by the satellites, so receivers can display current UTC or zone time, rather than the GPS system time. Currently there is a 11 second difference.

3.9.4 Differential GPS

Differential GPS (DGPS) is a method of correcting some system errors by using the errors seen at a known location to correct the readings of a moving receiver.

First the exact position of the reference station must be determined. As this reference station does not move its position can be compared with the calculated GPS position. The result is an error measurement which must be repeated because its value changes permanently. These error measurements are passed to the moving receiver which can adjust its GPS calculated position to compensate.

In addition to the permanent change of the error the measurement depends on the particular satellites used to compute the position. So the reference station must compute the errors in the pseudo range measurements for each single satellite which can be received. Thus the error in the pseudo range measurements, and other system status information to identify the satellites, must be broadcasted by some means (e.g. sub-carriers on FM). A differential GPS receiver receives and decodes this differential information. Then this information is combined with the pseudo range measurements from the GPS system before the receiver calculates the position.

DGPS will eliminate the error introduced by Selective Availability, and errors caused by variations in the ionosphere, resulting in reported positions within about 10 metres of the true position 95% of the time.

3.9.5 GNSS

3.9.5.1 Introduction

The evolution of GNSS is expected to occur in two phases. The first (GNSS-1) is a transitional step that builds upon the existing GPS and GLONASS satellite navigation systems. This phase will augment the current GPS and GLONASS navigation signals available to the public to provide improved availability, accuracy and integrity.

The European contribution to the second phase of the GNSS evolution (GNSS-2) will overcome many of the concerns regarding dependency upon third country systems. Examples of these are GPS and GLONASS.

3.9.5.2 GNSS-1

Europe's primary contribution to GNSS-1 will involve signal relay transponders carried on geostationary satellites, and a network of ground stations. Together they are to provide a regional augmentation service for GPS and GLONASS signals over Europe and many other parts of the world, known as the European Geostationary Navigation Overlay Service (EGNOS).

The EGNOS ground network will provide the backbone for two additional navigation services in addition to basic ranging, namely integrity monitoring and wide area differential corrections. The integrity service will allow range error estimates for each navigation signal to be broadcast, thereby enabling users to know within 10 seconds whether a navigation signal is out of tolerance. This is particularly important for safety critical applications, where action can be taken before any critical situation arises. EGNOS is due to reach full operational capability by 2003 and, when completed, will serve as a major regional component of what is hoped to be a seamless, world-wide navigation augmentation system.

3.9.5.3 GNSS-2 (GALILEO)

During 1999, the European Commission recommended that Europe should develop a new satellite navigation constellation known as Galileo. Galileo will be based on 21 or more medium Earth orbit satellites, being fully compatible with GPS, yet independent from it. The Galileo ground infrastructure will comprise a global network of monitoring stations and the associated system control and Earth stations. Galileo will provide to all users an accuracy of a few metres, which is beyond the current GPS standard positioning service.

Furthermore, Galileo will fulfil the requirements of safety critical applications by providing service guarantees, thereby enabling certifications and international standardisation. In addition to the basic service which will be available to everybody free of charge, a second service level or 'controlled access service' (CAS) will be offered to specific users, with associated availability and liability guarantees. Galileo will be fully operable in 2008, with signal transmission commencing in 2005.

3.10 Broadcast Radio

3.10.1 Introduction

The term “Broadcast Radio” is used to encompass radio transmissions that provide entertainment and other forms of listener services. Prime examples are the national broadcasting services that are provided in most Countries and funded centrally either by governments or by some kind of national licensing arrangements. However there are also now increasing numbers of commercial services that survive on the income from sponsorship or advertising.

Only three types of broadcast radio communications are presently used for the transmission of transport related data. The first two are Frequency Modulated (FM) and Digital Audio Broadcasting (DAB). In both cases, the facility to provide traffic and travel information is seen as an addition to the main purpose of the broadcasting service. The third type of communications (DARC) has yet to make significant inroads into the market, albeit that some new networks are being set up – it appears that DARC will be more appropriate for niche markets.

Table 1 Broadcast Radio Cuumincations - Available Types

Type	Data Rates	Mobile Use	Uni-/Bi-directional	Projected Life-span
DAB	1,7 Mbps	yes	uni-directional	>20 years
FM	1,1875 kbps	yes	uni-directional	another 10 years?
DARC	10 kbps	yes	uni-directional	another 10 years?

3.10.2 FM

There are two systems operating for data services on VHF/FM: RDS and DARC. The Radio Data System has been on the market since 1987 and is standardised by CENELEC (EN50067). Applications for RDS include allowing a receiver to search for the strongest frequency or for a particular programme within an transmitter network, the programme service name shown on an eight character display and travel information. The Traffic Message Channel (TMC) is the latest development using RDS.

3.10.3 DAB

3.10.3.1 Description

The EUREKA-147 DAB system is a novel digital sound broadcasting system intended to supersede the existing analogue Frequency Modulation (FM) systems operating in Band II. As a broadcasting system, it is unidirectional. It is designed as a rugged, yet highly spectrum and power efficient sound and data broadcasting system. It has been designed for terrestrial and satellite as well as for hybrid (co-channel satellite and terrestrial delivery of the same multiplex in the same geographical area) and mixed delivery (same multiplex on different frequencies in different geographical areas).

DAB is considered as the future of radio as it makes more efficient use of crowded airwaves and provides CD-quality sound that is more robust and noticeably better than an FM analogue broadcast. DAB broadcasts are virtually immune to interference and fading; programmes are not suddenly degraded or even lost when the car passes through a tunnel or under power lines.

One of the principal advantages of switching to DAB is that a single frequency called a "Multiplex" can carry a combination of stereo and/or mono audio programmes. DAB allows to go beyond audio and may provide some or even all of the Multiplex capacity to transmit data that is not related to audio programming

The DAB system is an open, stable, fully proven and standardised system which was developed by the EUREKA-147Consortium. It is normalised as a European Telecommunications Standard ETS 300 401 (see ETSI web site http://www.etsi.org). It was first published in 1994 and revised in 1998. This standard describes the DAB emission signal and defines DAB as a system to provide mobile, portable; and fixed reception. The standard is based on the overall system and service requirements adopted by the ITU-R Recommendations 774 and 789.

The DAB system is recommended on the world-wide level by the ITU-R 1114 , as Digital System A, for terrestrial and satellite delivery. The audio coding algorithm used by the DAB system has been subject to the standardisation process within the ISO/Moving Pictures expert Group (MPEG), see ISO/IEC 11172-3 and ISO/IEC 13818-3. The layered ISO open system interconnected model ISO 7498 has been used to the extent possible, and interfacing to information technology equipment and communications networks has been taken into account where applicable.

3.10.3.2 DAB Standards

Complementary ETSI standards are as follows[6]:

- EN 301 234: DAB Multimedia Object Transfer (MOT) protocol
- EN 300 797: DAB Distribution interfaces; Service Transport Interface (STI)
- EN 300 798: DAB Distribution interfaces; Digital baseband In-phase and Quadrature (DIQ) interface
- ETS 300 799: DAB Distribution interfaces; Ensemble Transport Interface (ETI)

DAB receiving equipment standards are adopted by TC 206 of CENELEC as follows:

- EN 50248: Characteristics of DAB receivers
- EN 50255: Digital Audio Broadcasting system - Specification of the Receiver Data Interface

Receivers are becoming available and although currently much more expensive than those currently available for FM, within a few years are likely to cost only marginally more.

DAB is now entering the implementation stage; there are large scale trials in eleven other European countries as well as in other parts of the world. DAB regular services are now available in Sweden and the UK, and Germany has announced regular services. Once regular services are introduced, the broadcasters cannot simply switch them off.

3.10.3.3 DAB Frequency Management Aspects

The EUREKA-147 System is capable of operating in any frequency band between 30 and 3000 MHz. The required spectrum planning parameters for the EUREKA System are well defined and are available in the public domain (see for example the ITU Special Publication on Terrestrial and Satellite Digital Sound Broadcasting to vehicular, portable and fixed receivers in the VHF/UHF bands, Geneva, 1995). The System has been implemented in various parts of the world at frequencies in the region of 50, 80, 100, 210-240, 600, 800, and 1452-1492 MHz.

[6] The Eureka 147 Project has also finalised a specification on the return link to be used to provide an interactive channel (PSTN; ISDN, GSM).

Generally, the higher the frequency, the higher the cost of the coverage. Also, building attenuation increases with the frequency. Nevertheless, all the experiments, and recent operational services in L-band) have shown that the EUREKA System performs very well at higher frequencies, including in L-Band. Lower frequencies such as Band I are characterised by a high level of man-made noise and should be avoided.

The introduction of DAB requires a forward-thinking long-term plan. To this end, sufficient radio-frequency spectrum is required to prevent significant co-channel and adjacent-channel interference between DAB services. Non-occupied spectrum is required in blocks of about 1.75 MHz bandwidth. In Europe, such spectrum slots are normally not available in Band II which is extensively used by FM services. Most of the countries opted to start their DAB services in Band III which is heavily used by terrestrial television or in L-Band which is used by fixed services, radio astronomy, mobile and telemetry / aero-navigational services in many countries.

A three-week Planning Meeting was convened in July 1995 by the European Conference of Postal and Telecommunications Administrations (CEPT). The aim of this Meeting was to produce a Special Arrangement for the introduction of terrestrial transmissions of the EUREKA-147DAB system in the frequency bands 46 - 68 MHz, 174 - 240 MHz and 1452 – 1467,5 MHz, as well as to prepare an associated Frequency Block Allotment Plan, taking into account the final requirements of the CEPT member countries.

The Allotment Plan drawn up at the Meeting provides practically all the member countries of the CEPT with two sets of frequency blocks, each of width 1,536 MHz. This is a vital prerequisite to the wide-scale launch of terrestrial DAB services in Europe. Most of the CEPT countries opted for frequency block allotments in VHF Band III and in L-Band. This table gives the overall number of all allotments agreed or subject to an agreement in all CEPT countries:

Table 2 Allocation of DAB Bands

Band	Frequency Range	No of Wiesbaden allotments	No of allotments with a reference to an agreement
Band I	47 – 68 MHz	1	0
Band II	87 – 108 MHz	1	0
Band III	174 – 240 MHz	278	205
Part of L-Band	1452 – 1467,5 MHz	467	177
Part of L-Band +	Above 1467,5 MHz	4	0
Total		**751**	**382**

Eighty-five frequency blocks can potentially be used for current and future DAB services in Europe. These blocks are located in VHF Band I (47 - 68 MHz), Band II (87 - 108 MHz), Band III (174 MHz - 240 MHz) and in part of L-Band (1452 - 1492 MHz). In September 1998, the CEPT identified a need for 7 additional frequency blocks at L-Band so that the total number of blocks available for terrestrial DAB at L-Band will be 16 (out of 23). This extension should allow for a third frequency block to become available at any given location.

It should be pointed out that the currently available DAB blocks are insufficient to accommodate all the existing national, regional and local broadcasters using FM and to covert them in DAB. To this end, further blocks will be needed. An attractive option, subject to agreement with the military, is the 230 – 240 MHz band (i.e. Channel 13).

All broadcasting bands are subjected to a regulation that defines the content, i.e., audio or data. Today, data services are allowed but the level varies from country to country depending of national regulatory broadcasting arrangements where available. However, the process of convergence of broadcasting and telecommunication already started in various countries thus paving the way towards full data broadcasting.

Each frequency block carries a two- or three-character label which is convenient for receiver manufacturers and consumers to use when initially programming their receivers. The labelling system of the frequency blocks in VHF Band I and Band III is fully compatible with the existing VHF television channel numbers (i.e. Channels 2 to 13). Each of these television channels can accommodate four DAB blocks; six blocks in the case of Channel 13.

3.11 Television

3.11.1 Overview

The use of television is currently confined to the dissemination of traffic and travel information. This is because at the moment there is no mechanism for its safe use in vehicles, and because it does not allow user (Traveller) interaction. This may change with the advent of Digital Video Broadcasting (DVB – see below).

3.11.2 Analogue

Analogue television has been used as a mechanism to disseminate of traffic and travel information for some time. This is particularly true in the UK, where both the British Broadcasting Corporation (BBC) and the Independent Television companies have both supported information systems called Ceefax and Oracle respectively.

Both the UK systems work by using “spare” parts of the analogue television signal that is used to transmit programmes. Television sets interrogate this signal and store the information that it contains. The television viewer can select “information pages” for display instead of the television programmes, and can navigate through the “pages”. Some “pages” are in fact multiple displays that scroll automatically. The scrolling process can be frozen, but parts of the scrolled information cannot be selected. Only a limited amount of data can be sent with each television screen update, so displaying the selected “page” can take a little while, whilst all the transmission goes through all the other “pages”. However only the selected “page” will be displayed, the screen remaining blank or showing the previous “page”, until the selected “page” is reached.

The information is generated by specialised systems run by the Broadcaster, and they are provided with the traffic and travel information through manual input. This is done so that the Broadcaster can exercise editorial control over what is displayed, and can ensure that the information is edited to fit within the “page” format. The “pages” display text and character based graphics, using a low resolution – about 20 lines with 30 characters per line. This makes the graphics rather "crude", but good enough to show an identifiable map of the UK.

Both Ceefax and Oracle provide comprehensive traffic and travel information. This can be selected by mode, and by UK region. It may also include severe weather warnings when appropriate. Because of the limitations of the "page" format, the information can be very concise which may leave the viewer with not much idea about the real situation. For example there are no maps to show the location of incidents within the road network. There is also no information about such things as parking.

As noted in the introduction, this system is not interactive because the viewer cannot select particular information through input. However the information is organised by mode (for the whole UK) and by UK region (covers road, ferry and train). It can also be time consuming because of the low refresh rate, which means that it may take several minutes to view the required "page(s)".

3.11.3 Digital Video Broadcasting (DVB)

3.11.3.1 Background

Until late 1990, digital television broadcasting to the home was thought to be impractical and costly to implement. During 1991, broadcasters and consumer equipment manufacturers discussed how to form a concerted pan-European platform to develop digital terrestrial TV. Towards the end of that year, broadcasters, consumer electronics manufacturers and regulatory bodies came together to discuss the formation of a group that would oversee the development of digital television in Europe - the European Launching Group (ELG).

The ELG expanded to include the major European media interest groups, both public and private, the consumer electronics manufacturers, common carriers and regulators. It drafted a Memorandum of Understanding (MoU) establishing the rules by which this new and challenging game of collective action would be played.

3.11.3.2 Memorandum of Understanding

The concept of the MoU was a departure into unexplored territory and meant that commercial competitors needed to appreciate their common requirements and agendas. Trust and mutual respect had to be established. The MoU was signed by all ELG participants in September 1993, and the Launching Group became DVB (Digital Video Broadcasting). Work to develop digital television, already underway in Europe, moved into top gear. Around this time, the Working Group on Digital Television prepared a study of the prospects and possibilities for digital terrestrial television in Europe. The highly respected report introduced important new concepts, such as proposals to allow several different consumer markets to be served at the same time (e.g. portable television and HDTV). In conjunction with all this activity, change was coming to the European satellite broadcasting industry, and it was becoming clear that the once state-of-the-art MAC systems would have to give way to all-digital technology. DVB provided the forum for gathering all the major European television interests into one group. It promised to develop a complete digital television system based on a unified approach. It became clear that satellite and cable would deliver the first broadcast digital television services. Fewer technical problems and a simpler regulatory climate meant that they could develop more rapidly than terrestrial systems. Market priorities meant that digital satellite and cable broadcasting systems would have to be developed rapidly. Terrestrial broadcasting would follow.

3.11.3.3 DVB Standards

By 1997 the development of the DVB Project had successfully followed the initial plans. The project had also entered its next phase, promoting its open standards globally, and making digital television a reality.

Open, Interoperable, Flexible, Market-led, Open

DVB systems are developed through consensus in the working groups of the Technical Module. Members of the groups are drawn from the general assembly of the project. Once standards have been published, through ETSI, they are available at a nominal cost for anyone, world-wide. Open standards free manufacturers to implement innovative and value added services. It doesn't matter where DVB technology is developed. It is available world-wide.

Interoperable

Because the standards are open, all the manufacturers making compliant systems are able to guarantee that their DVB equipment will work with other manufacturers' DVB equipment. Not only this, but because the standards are designed with a maximum amount of commonality, and based on the common MPEG-2 coding system, they may be effortlessly carried from one medium to another, which is frequently needed in today's complex signal distribution environment. DVB signals move easily and inexpensively from satellite to cable, from cable to terrestrial.

Flexible

Owing to the use of MPEG-2 packets as "data containers", and the critical DVB Service Information surrounding and identifying those packets, DVB can deliver to the home almost anything that can be digitised. It does not matter whether this is High Definition TV, multiple channel Standard definition TV (PAL / NTSC or SECAM) or even exciting new broadband multimedia data and interactive services.

Market-led

In contrast to earlier initiatives in Europe and the United States, the DVB Project works to strict commercial requirements established by organisations who work every day to meet its needs. It is not a regulator or government-driven (top-down) initiative. Working to tight timescales and strict market requirements means achieving a considerable economy of scale, which ensures that, in the transformation of the industry to digital, broadcasters, manufacturers and, finally, the viewing public will benefit.

3.11.3.4 DVB Standards Adoption

For each specification, a set of User Requirements is compiled by the Commercial Module. These are used as constraints on the specification. User requirements outline market parameters for a DVB system (price-band, user functions, etc.).

The Technical Module then develops the specification, following these user requirements. The approval process within DVB requires that the Commercial Module supports the specification before it is finally approved by the Steering Board.

Following approval by the Steering Board, DVB specifications are offered for standardisation to the relevant international standards body (ETSI or CENELEC), through the EBU/ETSI/CENELEC JTC (Joint Technical Committee), the ITU-R, ITU-T and DAVIC.

3.11.3.5 DVB – The Future

At the moment DVB carries digital versions of the analogue channels, plus other specialised digital only channels. Unlike analogue television, it is only available through subscription, although some hotels may provide access to some channels as part of "room service". Also there is no dedicated traffic and travel information channel at the moment. It would probably have to be provided as part of another channel, or "bundled in" with the price of another channel. There should also be scope for viewer interaction via the "set top" boxes that DVB uses for channel selection, etc. This means that DVB is becoming more like an Internet service, which may appeal to those who do not want to have a home PC with an Internet connection and opens up the possibility of other services such as personalised trip planning being provided.

3.12 Short Range Communication Systems

3.12.1 Introduction

There are two short range communications systems that are available for use in ITS. These comprise Bluetooth, a short range radio link, and the DSRC standards being developed in conjunction with the automotive industry.

3.12.2 Bluetooth

Bluetooth is the codename for a technology specification for small form factor, low-cost, short range radio links between mobile PCs, mobile phones and other portable devices.

It will enable users to connect a wide range of computing and telecommunications devices easily and simply, without the need to buy, carry, or connect cables. It delivers opportunities for rapid ad hoc connections, and the possibility of automatic, unconscious, connections between devices. It will virtually eliminate the need to purchase additional or proprietary cabling to connect individual devices. Because Bluetooth can be used for a variety of purposes, it will also potentially replace multiple cable connections via a single radio link.

Bluetooth will offer a short range communication at ranges of about 10 meters even if the terminals are on the move or when there is no line-of-sight between those terminals. The system will operate at the unlicensed spectrum area close to 2,45 Ghz. The expected data transmission speeds are between 720 kbps and one Mbps.

The transmission is secured by built in encryption and authentication methods. In addition a frequency-hopping scheme with 1600 hops/sec is employed. All of this together with an automatic output power adoption to reduce the range exactly to requirement makes the system difficult to eavesdrop.

3.12.3 DSRC

Dedicated Short Range Communication (DSRC) is a common component of practically all existing Electronic Fee Collection (EFC) systems. Its ability to support other ITS services such as Traffic and Travel Information (TTI) and access control has also been widely demonstrated.

The 5.8 GHz frequency band is allocated for short range communication. The standardisation of short range communication is already far advanced and several pre-standards have already been approved. The European standardisation body CEN has a specific technical committee assigned to the issues of road transport and traffic telematics, namely CEN TC 278.

The working groups of TC 278 have elaborated a set of pre-standards which may be amended in the next few years before they will be changed to proven standards. There are approved pre-standards for some parts of the DSRC specification. These comprise the following:

- layer 1, defining the physical characteristics of the DSRC interface;
- layer 2, describing the procedures for transmission medium access and for logical link control between the Road Side Equipment (RSE) and the On Board Equipment (OBE);
- layer 7, defining the basic application services that the DSRC communication offers to all the specific telematics services that want to make use of the DSRC communication stack.

For EFC, there is already a pre-standard in place, the Application Interface Definition for EFC, that defines the structure of the communication as well as the functions and data elements available to perform an EFC transaction between the RSE and the OBE via the DSRC link. Although a lot of standardisation work has been done, it has to be highlighted that these standards are only enabling interoperability but not providing interoperability automatically.

4 WIRED TECHNOLOGIES

4.1 Introduction

Wired technologies cover those that use some form of fixed link. Typically this may be copper or fibre, but other media may also be used, or be developed in the future.

4.2 Telephone

4.2.1 Introduction

The main task of Telephony is to give the customer the possibility to carry on a speech conversation with someone far away. For this electrical signals are transmitted by wires between the locations of these customers. By the time several data services evolved.

The early phone network consisted of a pure analogue system that connected telephone users directly by an interconnection of wires. This system was very inefficient, was very prone to breakdown and noise, and did not lend itself easily to long-distance connections. Beginning in the 1960s, the telephone system gradually began converting its internal connections to a digital switching system. Today, nearly all voice switching is digital within the telephone network. Still, the final connection from the local central office to the customer equipment is an analogue Plain-Old Telephone Service (POTS) line. But more and more of these connections are replaced by digital (e.g. ISDN) lines.

4.2.2 POTS (analogue)

POTS (Plain Old Telephony Service) stands for the commonly known analogue wireline telephone service. It has provided the main form of communication link between the “central” and “roadside” parts of ITS applications for over thirty years. Many applications still use it because there is a large infrastructure available that (for the moment) reaches many geographic locations that digital networks do not reach. The technology required to transmit the data is relatively simple (modems), cheap to purchase and widely available from many suppliers. However there are several factors that are forcing ITS applications to move to other forms of wired links. These can be summarised as follows:

- high line costs compared with the amount of data that can be transferred;
- low capacity in terms of the amount of data that can be transmitted in a given time, particularly if services providing travel information such as graphics are required;
- increasing availability of relatively cheap digital based links due to other pressures, e.g. the drive for more widely available high speed access to the Internet.

Therefore although POTS is currently widely used, it is expected that in the future the trend will be to other forms of communication. Thus there may be little or no future development of new equipment that uses POTS. However the existing equipment will continue to be available from suppliers and used for some time (years) to come. One reason for this will be its wide spread availability in less developed countries and its ability to use low quality physical links.

4.2.3 ISDN

ISDN, which stands for **I**ntegrated **S**ervices **D**igital **N**etwork, is a system of digital phone connections which has been available for some time. This system allows data to be transmitted using end-to-end digital connectivity.

With ISDN, voice and data are carried by bearer channels (B channels) occupying a bandwidth of 64 Kbps. A data channel (D channel) handles signalling at 16 Kbps or 64 Kbps, depending on the service type.

There are two basic types of ISDN service: Basic Rate Interface (BRI) and Primary Rate Interface (PRI). BRI consists of two 64 kbps B channels and one 16 kbps D channel for a total of 144 kbps. This basic service is intended to meet the needs of most individual users or the SOHO market.

PRI is intended for users with greater capacity requirements. In Europe, PRI consists of 30 B channels plus one 64 kbps D channel for a total of 1984 kbps. It is also possible to support multiple PRI lines with one 64kbps D channel using Non-Facility Associated Signalling (NFAS).

To access BRI service, it is necessary to subscribe to an ISDN phone line. Customer must be within 5,5 km of the telephone company central office for BRI service; beyond that, expensive repeater devices are required, or ISDN service may not be available at all. Customers will also need special equipment to communicate with the network switch and with other ISDN devices. These devices include ISDN Terminal Adapters and ISDN Routers.

Speed

ISDN allows multiple digital channels to be operated simultaneously through the same regular phone wiring used for analogue lines. The change comes about when the telephone company's switches can support digital connections. Therefore, the same physical wiring can be used, but a digital signal, instead of an analogue signal, is transmitted across the line. This scheme permits a much higher data transfer rate than analogue lines. BRI ISDN supports an uncompressed data transfer speed of 128 kbps. In addition, the latency, or the amount of time it takes for a communication to begin, on an ISDN line is typically about half that of an analogue line. This improves response for interactive applications.

Multiple Devices

With POTS (see chapter above), it was necessary to have a phone line for each device you wished to use simultaneously. For example, one line each was required for a telephone, fax, computer, bridge/router, and live video conference system. Transferring a file to someone while talking on the phone or seeing their live picture on a video screen would require several phone lines.

It is possible to combine many different digital data sources and have the information routed to the proper destination. Since the line is digital, it is easier to keep the noise and interference out while combining these signals. ISDN technically refers to a specific set of digital services provided through a single, standard interface. Without ISDN, distinct interfaces are required instead.

Signalling

Instead of the phone company sending a ring voltage signal to ring the bell in the phone ("In-Band signal"), it sends a digital packet on a separate channel ("Out-of-Band signal"). The Out-of-Band signal does not disturb established connections, and call set up time is very fast.

A multitude of supplementary services is defined by ISDN, too. Some of these services are the indication who is calling, what type of call it is (data/voice/fax), and what number was dialled. These supplementary services are for example used by intelligent ISDN phone for making decisions on how to direct the call.

A special packet switched data service (called X.31) is defined within ISDN. This service uses the D-channel to transmit packet data to the X.25 network. Thus this service will be described in that chapter.

4.3 DSL

4.3.1 Introduction

The Digital Subscriber Line (DSL) was developed to bring high-bandwidth information to homes and small businesses over ordinary copper telephone lines. Hence, it allows to use the existing infrastructure to add data transmission on wires which were only transporting voice up to now. The purpose of these technologies is mainly to provide permanent connections to a data network at high rates, with limited costs. In other words, They are mainly high data rates local loop technologies. Commercial offers are just beginning to appear in Europe.

4.3.2 Types of DSL

The term xDSL refers to different variations of DSL, such as ADSL, HDSL, and RADSL. The speed of an xDSL line can go up to 6,1Mbps (of a theoretical 8,448Mbps). More typically, individual connections will provide from 1.544Mbps to 512Kbps downstream (from the network to the user) and about 128Kbps upstream (from the user to the network). A DSL line can carry both data and voice signals. The part of the line dedicated to data is continuously connected. Compaq, Intel, and Microsoft working with telephone companies have developed a standard and easier-to-install form of ADSL called G.Lite that is expected to accelerate deployment. Within a few years, DSL is expected to replace ISDN in many areas and to compete with the cable modem in bringing high-speed access to homes and small businesses.

4.3.3 Overview Description

DSL is very different from traditional use of a modem on a telephone line. With a traditional modem. the ability of a home PC to receive information has to go through a bottle neck: the telephone company filters information that arrives as digital data, puts it into analogue form into telephone line, using the same narrow channel as that used for voice. In other words, the analogue transmission between your home or business and the phone company is a bandwidth bottleneck.

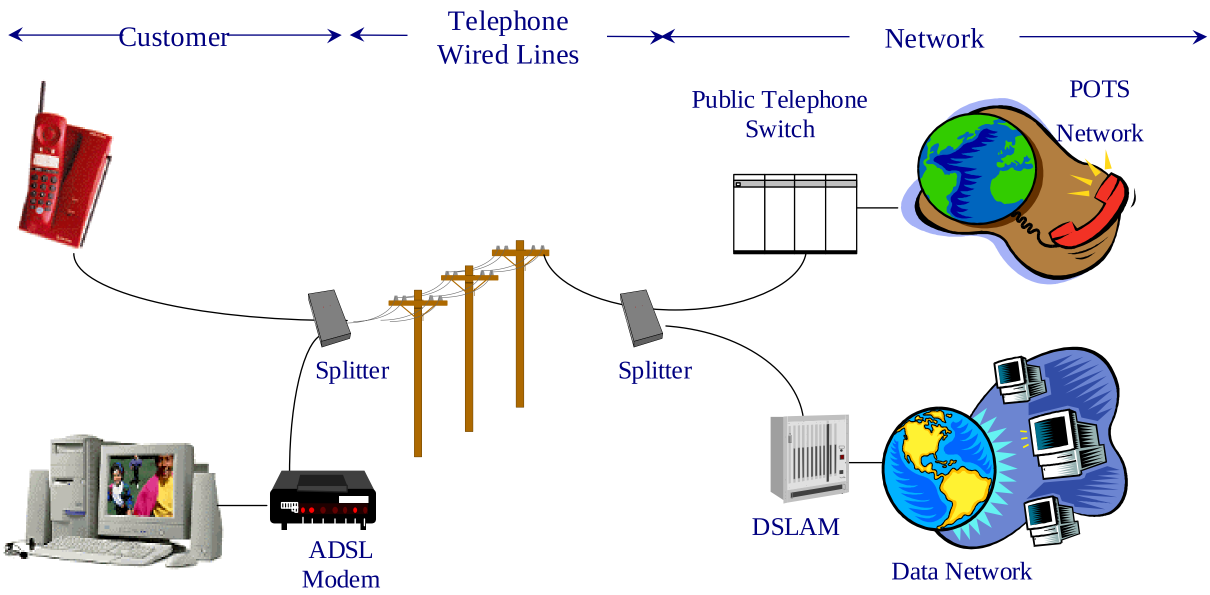
DSL assumes digital data does not require conversion to analogue voice transmission. Digital data is transmitted to your computer as-is and this allows the phone company to use a much wider bandwidth for transmitting it to you. By the way, the signal can be separated so that some of the bandwidth is used to transmit an analogue voice signal. This enable to use the telephone and computer on the same line and at the same time. Different DSL modem makers are using either Discrete Multitone Technology (DMT) or Carrierless Amplitude Modulation (CAP). A third technology, known as Multiple Virtual Line (MVL), is another possibility that may be used.

To interconnect multiple DSL users to a high-speed backbone network, the telephone company uses a Digital Subscriber Line Access Multiplexer (DSLAM). Typically, the DSLAM connects to an ATM network that can handle data transmission at gigabit data rates. At the other end of each transmission circuit, the reverse operation is made: a DSLAM demultiplexes the different signals and forwards them to appropriate individual DSL end user.

Another issue is linked to the splitter. In order to separate data from voice, most DSL technologies require that a signal splitter is installed at the user end, causing the expense of an installation by a professional. However, it is possible to avoid this using a specific standard. This is known as *splitterless* DSL, DSL Lite, G.Lite, or Universal ADSL and has recently been made a standard.

4.3.4 ADSL.

This variation (Asymmetric Digital Subscriber Line) is the form of DSL that will become most familiar to home and small business users (SOHO – Small Office/Home Office). ADSL is called "asymmetric" because most of its two-way or duplex bandwidth is devoted to the downstream direction, sending data to the user. Only a smaller portion of the bandwidth is available for upstream, form the user. This is was designed to fit the Internet model. Most Internet and especially graphics- or multi-media intensive Web data need lots of downstream bandwidth, but user requests and responses are small and require little upstream bandwidth. Using ADSL, up to 6.1 megabits per second of data can be sent downstream and up to 640 Kbps upstream. The high downstream bandwidth means that your telephone line will be able to bring motion video, audio, and 3-D images to your computer or hooked-in TV set. In addition, a small portion of the downstream bandwidth can be devoted to voice rather data, and you can hold phone conversations without requiring a separate line.



4.3.5 CDSL.

Consumer DSL is a trademarked version of DSL that is somewhat slower than ADSL (1 Mbps downstream, probably less upstream) but has the advantage that a "splitter" does not need to be installed at the user's end. CDSL uses its own carrier technology rather than DMT or CAP ADSL technology.

4.3.6 G.Lite or DSL Lite.

This is also known as DSL Lite, splitterless ADSL, and Universal ADSL. It is essentially a slower ADSL that doesn't require splitting of the line at the user end but manages to split it for the user remotely at the telephone company. This saves the cost of installing a splitter at home. G.Lite provides a data rate from 1.544Mbps to 6Mbps downstream and from 128 Kbps to 384 Kbps upstream. G.Lite is expected to become the most widely installed form of DSL.

4.3.7 HDSL.

The earliest variation of DSL to be widely used has been HDSL (High bit-rate DSL) which is used for wide band digital transmission within a corporate site and between the telephone company and a customer. The main characteristic of HDSL is that it is symmetrical: an equal amount of bandwidth is available in both directions. For this reason, the maximum data rate is lower than for ADSL. HDSL can carry as much on two wires of twisted-pair as can be carried on a T1 line in North America or an E1 line in Europe (2,320 Kbps). This technology has several drawbacks, in particular two wires are needed, and it is quite “noisy”. For this reason a second standard, HDSL 2 should appear beginning of 2000.

4.3.8 S-HDSL.

Single pair HDSL operates on a single copper pair, as opposed to HDSL requiring two and allows operators to implement high symmetrical bit rates on a single local loop while maintaining the existing telephone service on the same line. The typical bit rate is 768kbps.

4.3.9 IDSL.

This is a hybrid combination of ISDN and ADSL, essentially being a form of ISDN where the data signal is separated before the switch. Currently, IDSL supports speeds of 128kbps. Newer IDSL modems will take advantage of the unused D-channel bandwidth and support 144kbps connections. In the European market, ISDN seems to have taken over and IDSL is unlikely to make a substantial impact.

4.3.10 RADSL.

Rate-Adaptive DSL is an ADSL technology designed to operate at a variety of data rates. Software is able to determine the rate at which signals can be transmitted on a given customer phone line and adjust the delivery rate accordingly. Westell's FlexCap2 system uses RADSL to deliver from 640 Kbps to 2.2 Mbps downstream and from 272 Kbps to 1.088 Mbps upstream over an existing line.

4.3.11 SDSL.

Single-line DSL is apparently the same thing as HDSL with a single line, carrying 1.544 Mbps (U.S. and Canada) or 2.048 Mbps (Europe) each direction on a duplex line.

4.3.12 UDSL.

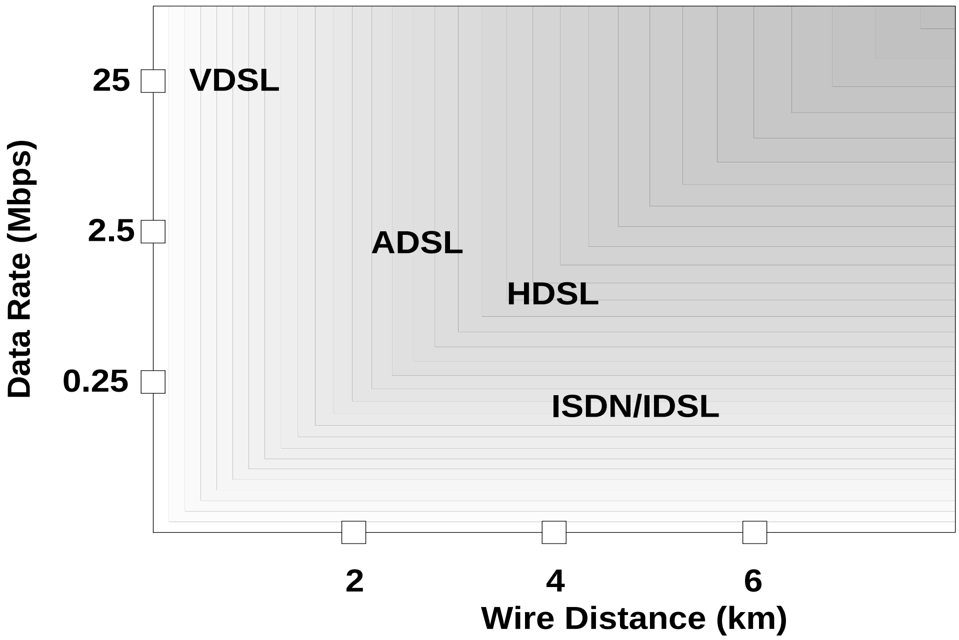
Unidirectional DSL is a proposal from a European company. It's a unidirectional version of HDSL.

4.3.13 VDSL.

Very high data rate DSL is a developing technology that promises much higher data rates over relatively short distances (between 51 and 55 Mbps over lines up to 1,000 feet or 300 meters in length). It's envisioned that VDSL may emerge somewhat after ADSL is widely deployed and co-exist with it. The transmission technology (CAP, DMT, or other) and its effectiveness in some environments is not yet determined. A number of standards organisations are working on it.

The actual data rate of connections with the various xDSL depends on several factors. In particular, surrounding electric “noise”, physical quality of the wires, bridge taps, and especially distance. In fact, each technology can only be used on a limited distance as shown on the following chart.

Figure 4 xDSL speed vs copper pair length



4.4 X.25 (Packed Switched Data)

In the 70s, the need for interconnecting international networks has arisen. In response, telecommunication operators have developed the X.25 protocol.

X.25 is a low level network protocol based on packet switching. Packet-switched networks enable end stations to dynamically share the network medium and the available bandwidth. Variable-length packets are used for more efficient and flexible transfers. These packets then are switched between the various network segments until the destination is reached. Statistical multiplexing techniques control network access in a packet-switched network. The advantage of this technique is that it accommodates more flexibility and more efficient use of bandwidth.

X.25 implements the 3 lowest layers of the OSI model. It is based on Data Terminal Equipment (DTE) at end users premises or at the edge of the network and a meshed network of nodes equipment (called Data Circuit Terminating). A DTE user on an X.25 network may communicate with a number of remote DTEs simultaneously.

It allows international secure transmission of data, with error correction and flow control at each node. The transmission delay depends on the number of nodes between DTE. On most single networks the turn-around delay is about 0.6 seconds. This makes it unsuitable for voice telecommunication, and may be a serious drawback for transaction based on heavy interactions.

Prior to a transmission, a virtual circuit (VC) must established between the communicating DTE. Permanent virtual circuit (PVC) can also be set up. The integrity of the delivered data is guarantied. Packets are received in the same order they were sent.

It is also possible to send datagramms, which contain the address of the receiver DTE. In that case there is no need to establish a virtual circuit, but the delivery is not guarantied and they are received in any order.

In most parts of the world, X.25 is paid for by a monthly connect fee plus packet charges. There is usually no holding charge, making X.25 adequate for permanent connections. Another useful feature is speed matching: because of the store-and-forward nature of Packet Switching, plus excellent flow control, DTEs do not have to use the same line speed. So it is possible to have, for instance, a host connected at 56kbps communicating with numerous remote sites connected with cheaper 19.2kbps lines.

4.5 Frame Relay

Frame Relay is a high-performance WAN protocol that operates at the physical and data link layers of the OSI reference model (the two lowest layers). Like X.25, Frame Relay is an example of a packet-switched technology (see above). Frame Relay transports variable size frames (e.g. IP packets) through established virtual circuits (VCs) making use of statistical multiplexing.

Compared to X.25, Frame Relay reduces the packet processing at the internal network node, since the error control/retransmission and flow control is performed on an end to end basis at the network edge nodes. On the other hand, it does not provide guaranty on data integrity. The network delivers frames, whether the CRC[7] matches or not. It does not even necessarily deliver all frames, discarding frames whenever there is network congestion. Thus it is imperative to run an upper layer protocol above Frame Relay that is capable of recovering from errors, such as IPX or TCP/IP. In practice, however, the network delivers data quite reliably. Unlike the analogue communication lines that were originally used for X.25, modern digital lines have very low error rates. Very few frames are discarded by the network, particularly at this time when the networks are operating at well below design capacity.

Frame Relay offers a VPN service through the use of permanent VCs. Although switched virtual circuit specifications are available from the Frame Relay Forum, existent Frame relay networks provide only permanent virtual circuits. Upon establishment of a virtual circuit, the user specifies a CIR (Committed Information Rate) that must bee guaranteed by the network provider. Current CIR values range from 64 kb/s up to 2 Mb/s and limited (in specific cases) to 45 Mbps. It was proposed by the Frame Relay Forum. Commercial offers allow to aggregate several circuits to obtain higher throughput (typically up to 8 Mb/s).

Frame Relay is used mostly to route Local Area Network protocols such as IPX or TCP/IP. It can also be used to carry asynchronous traffic, SNA or even voice data. With respect to voice data, voice is digitised and compressed; silent periods (and echo) are omitted. Every frame can transport several voice packets. Voice is compressed at 6,3 kbps, 8 kbps or 16 kbps. For example, a 64 kbps link can simultaneously transport up to six 8 kbps (+ overhead) compressed communications. Voice over frame relay is seen as an attractive solution for

[7] CRC (Cyclic Redundancy Check): for each frame a CRC is calculated based on all of its bits. If the frame has been altered during transmission, the receiver will calculate a different CRC which enables him to notice the alteration, and, under certain conditions, to correct it.

companies requiring voice and data integration. Typical European Frame Relay networks have an ATM core network with Frame relay links (from 64 kbps up to 2 Mbps).

4.6 ATM

ATM is a high speed switching technology that allows the integration of different traffic media over a single physical network.

ATM was adopted by the ITU-T as the standard for the B-ISDN. Nevertheless, there are few similarities with ISDN, and standards are now mostly provided by the ATM Forum. With ATM all data (digital voice, video, files, etc.) are transferred in small fixed length cells (53 bytes, 5 bytes of overhead) over virtual connections. Connections are established prior to any data transfer.

This technology has been defined for use over wired lines, and enables high data rates. Instead of direct use over wires, it may also be installed over high speed SDH networks[8], or even WDM networks[9]. It may provide network services in direct connection with applications (e.g. to transport voice or video), in particular for large Local Area Networks, but the most common place of implementation is the backbone of operators' networks, in particular it is adapted to the transport X.25, Frame Relay or IP data.

At connection establishment phase, the user selects the type of the required connection. According to the Quality of service needed to transfer his information several kinds of services are available, in particular the amount of bandwidth needed and the shape of the traffic is negotiated prior to any connection.

Since all kinds of data flows may be sent on the same wires through switched cell flows, ATM statistical multiplexing improves resource utilisation. The granularity of transmission link speeds and the support of QoS make it suitable for high-speed applications having different characteristics.

Standardisation bodies (ITU-T, IETF, ATM Forum) propose solutions for scalable signalling and routing solutions such as PNNI, or IP over ATM support such as MPLS, Classical IP over ATM (CLIP, RFC 1577), LANE, or MPOA.

ATM is currently widely used in backbone networks while other high-speed technologies (e.g. switched Ethernet, Gigabit Ethernet) appear as an alternative option at the LAN level. Industries with stringent QoS and bandwidth requirements see ATM as an attractive (and sometimes unavoidable) solution.

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[8] SDH (Synchronous Data Hierarchy): protocol of high speed low level network that enables high speed transport of other protocols.

[9] WDM (Wavelength Digital Multiplexing): protocol for very high speed low level networks that make use of several wavelengths over fibber optical networks instead of a traditional single wavelength. It enables very high speed transport of other protocols.

4.7 Internet

4.7.1 Overview

The use if the Internet is becoming more widespread, both in terms of take-up by users and in terms of the information that is available. Until very recently this has been based on “wired” links, with “wireless” links being confined to access from a PC (usually a Laptop) via a mobile telephone. This is not very common as it can be very expensive.

However there is an increasing move towards the development of Internet access through “wireless” based technology, such as WAP. Its “wired” form is considered here, as the WAP version is still largely under development within Europe. However it should probably be considered as the Internet access mechanism of the future.

4.7.2 Internet Services through “wired” links

Internet Services are provided through Service Providers. These enable those with home or office PC’s (including Laptops) to have access to Web Sites. These Web Sites are used to display a wide variety of information in varying amounts of detail. It is also possible to provide interactive Web Sites so that the user can select exactly the information that is required. The content can include text and graphics, and can be of very high resolution. It is also possible to include video images, such as those provided by CCTV systems. For traffic purposed, this can include “real time” views of the actual traffic situation where ever there is a camera.

It would be impossible to describe the wide range of traffic and travel related Web Sites that are currently available. They also cover most modes of transport, particularly road, rail and Public Transport. For road information, many of the larger Cities and Towns throughout Europe are beginning to set up their own Web Sites to display the traffic and travel information. Generally they will show the information in various ways, that may include, graphical maps, and/or textual data, and/or video images. The maps vary from the rough schematic, to data overlaid on top of digital maps provided by specialist companies. Some sites also include trip planning, which may go outside the immediate area served by the operator of the Web Site.

Most users will access Web Sites from home or office based PC’s. Therefore their use is to gain an indication of the current traffic and travel situation. Unless access can be gained from a Laptop PC via a mobile telephone, the information that is provided will become increasing less useful to the Traveller once they have leave home or office (and the PC). This is because a great deal of traffic and travel information is usually time dependent and therefore becomes out of date.

4.7.3 Internet Services through WAP

As noted above, this is an area of intense development. It is therefore considered imprudent to try and describe what is available. However it is expected that for the Traveller, it will mean that information will be available whilst “ on the move”, making it possible to change travel plans as traffic and travel conditions change.

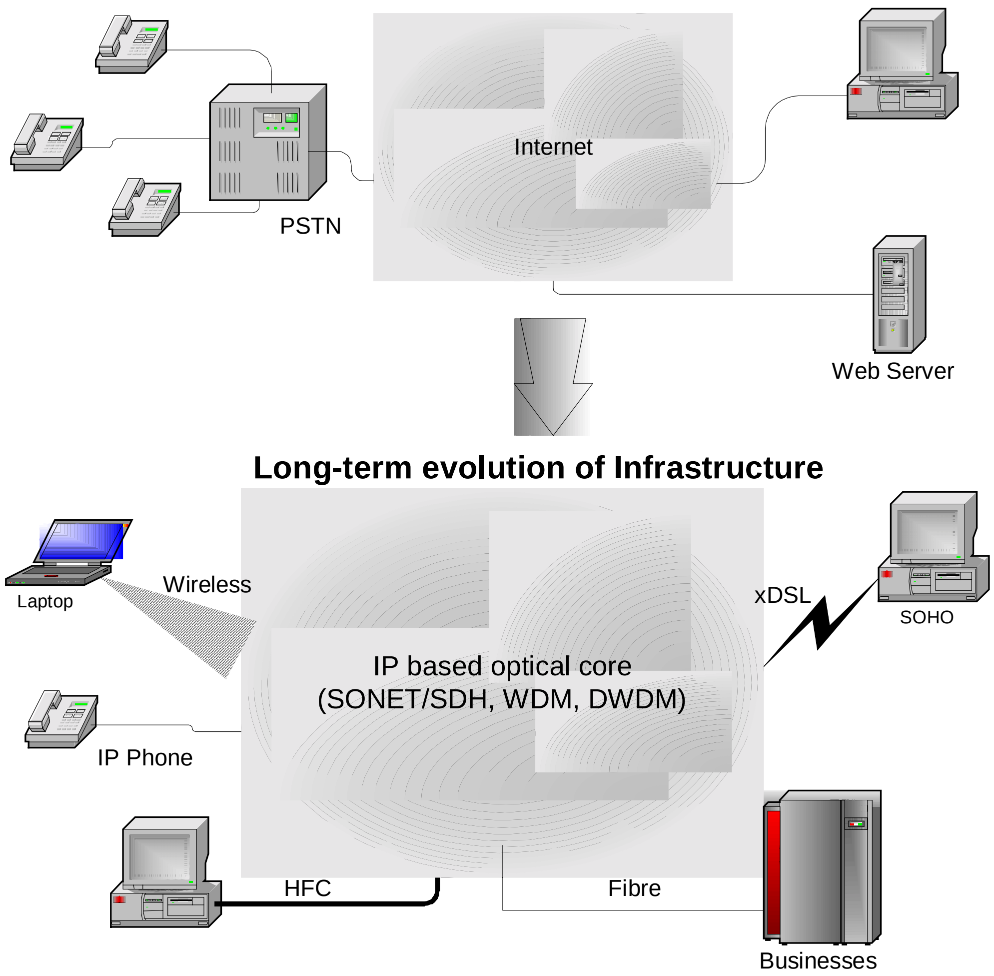
5 OPTICAL NETWORKING

5.1 Introduction

Below are described the two main techniques for a more efficient backbone through utilisation of optical fibres which may be more expensive to lay out but undoubtedly the fastest means available. SONET and WDM are discussed. There was a proposition by key companies in European backbone technology to move from ATM based to SONET based backbone since for the former, nearly 10% of traffic is overhead (5 bytes of header every 53 bytes of information). Encapsulating IP packets over ATM grows the overhead to 30% of the traffic whereas for SONET this falls down to less than 1%.

Currently, many ATM proponents argue that the lack of QoS in SONET cancel out this gain but the fact is that carrying IP over a SONET backbone has advantages that will grow as technology changes, like IP v6 for example.

Figure 5 Evolaution of Infrastructure

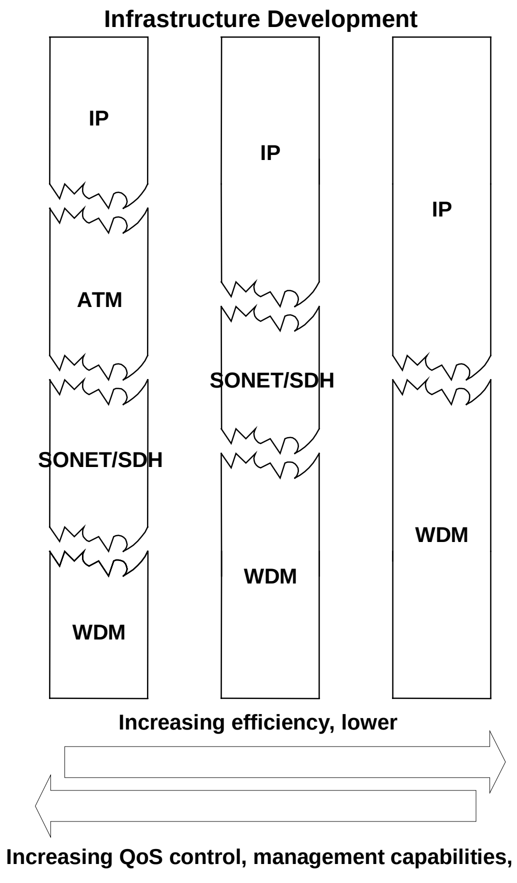


5.2 SDH/SONET

The Synchronous Digital Hierarchy/Synchronous Optical NETwork provides an advanced physical layer multiplexing transmission technology over fibre optic support. SDH/SONET provides flexible insertion/extraction of virtual circuits in a frame based hierarchical structure allowing transmission at high-speed rates.

Normalised speeds for SDH are 155,52 Mbps for STM-1 and up to 2488,32 Mbps for STM-16. SONET (US ANSI equivalent), provides an additional lower rate, OC-1 at 52,840 Mbps. User information is copied asynchronously into the user field of an SDH frame. A set of pointers (described in the header of the frame) allow to locate the user information within the frame allowing to directly extract, for example, a STM-1 circuit from a higher hierarchical multiplex such as a STM-16. SDH in ring configuration has the self-healing characteristic, restoring link failures in around 50ms. SDH/SONET currently appears as the most widely physical layer backbone technology deployed for transporting different types of upper layer traffic such as ATM, or even IP packets as described hereafter.

Figure 6 Comaprison of different types of SDH/SONET



5.3 Packet over SONET (POS)

This consists of encapsulating IP packets over PPP over SDH/SONET providing a gain in the order of 25% to 30 % compared to the IP/AAL5/ATM/SDH approach. Network switches and routers are also less expensive than ATM equipment. Nevertheless, POS does not have support for QoS and as IP traffic increase, IP routers may become traffic bottlenecks. POS equipment in the market can provide 155 Mbps (STM-1), 622 Mbps (STM-4) and 2,5 Gbps (STM-16) speeds. In POS networks, IP addresses are assigned to each interface and protocols such as RIP or OSPF are defined just as with any other IP network (in contrast, ATM requires address translation and higher management capabilities).

This technology is more oriented to operate over backbones dedicated to data transfer over the Internet while ATM is more adapted to multimedia developments (ATM will likely continue to appeal big telecom operators). The emergence of new protocols such as IPv6, RSVP, DiffServ, providing a certain degree of QoS, will probably favour this technology.

In order to increase capacity even more there are three distinct options for an optical network:

- Installation of new fibre equipment and transmission (expensive and time consuming)
- Transmission speed increase. Usual is 2.5Gbps (OC-48) and to extend beyond that, higher grade fibre is needed. Transmission at 10Gbps (OC-192) is starting to become an option and there is research for 40Gbps (OC-768).
- Use of Wavelength Division Multiplexing with the existing fibre structure.

The last mentioned of these seems appealing at the moment.

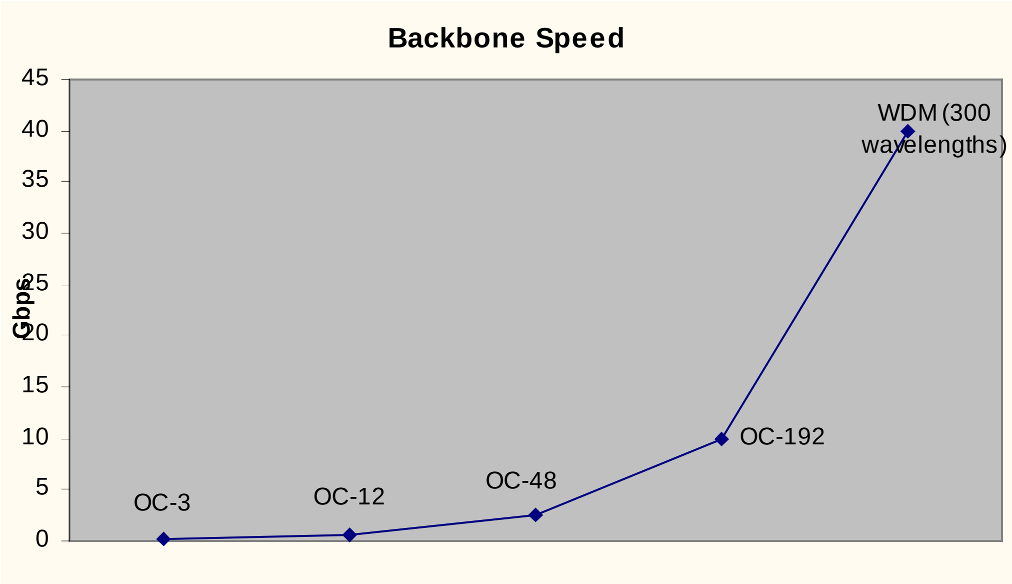
5.4 WDM

Wavelength Division Multiplexing and Dense-WDM (DWDM) technologies allow the multiplexing of several different wavelengths over a single fibre optic increasing the available bandwidth capacity of the fibre. The WDM technologies allow multiple lasers carrying independent signals to be transmitted in the same fibre with appropriate spacing of wavelengths (frequencies) between laser carriers.

Because of the broad bandwidths of lasers, the carried signals can be format-independent and be routed in networks independently. The lasers can carry digital or analogue signals. The digital signals can be any speeds (155MB/s *OC-3*, 625MB/s *OC-12*, 2.5GB/s *OC-48* or 10GB/s *OC-192*) or format (security-coding, compressed data and etc). The analogue signals can be any formats or co-exist with digital signals. The analogue signals can also be Frequency-Division-Multiplexed (FDM) in one laser because of the broad bandwidth of the laser beam.

For digital signals, WDM systems increase network capacities. For example, the maximum network capacity for a eight-wavelength WDM system increases to 20GB/s when each laser carries a 2.5GB/s (SONET OC-48) signal. Furthermore, the broadband feature of optical communications allows transparent networks where the lasers can carry independent signal formats. For example, wavelength_1 can carry analogue signals, wavelength_2 can carry a 2.5-GB/s digital signal while wavelength_3 carries a 155-MB/s digital signal. The transparent feature provides securities and flexibility to the networks. To achieve the transparency, the devices used in the networks have to be all-optical.

Figure 7 Graph of network capacity for different WDM sppeds



Currently, ITU-T has defined around a hundred different wavelengths, each of these wavelengths having a bandwidth of 2,5Gbps (up to 10Gbps in some cases, or more). WDM systems are universal and transparent to the type of flow they transport (e.g. digital signals, SDH, analogue signals such as video).

6 Synthesis Table

6.1 Introduction

The table in the next section shows the main characteristics of the Telecommunication Technologies linked with ITS. The meanings of the columns are summarised in the table below:

Table 3 Values of the Synthesis Table

Item	Sub-Item	Values	Description
Technology			Name of the Technology (Communication System)
Name			Name of a special Supplementary Service of that technology.
Configuration	Type	*x*-*y*	*x* and *y* are chosen among p, m, and b. p = point; m = multi-point; b = broadcast. Hence p-m = point to multi-point configuration. This means that the *same message* is send from one point and received simultaneously at several points.
	Connectivity (Con)	Wl, Wd	Wl = wireless wd = wired technology.
	Range	Short, medium long	(this may be only applicable for wireless)
	Directions (Dir)	d, sd, u	d = duplex, sd = semi duplex and u = unidirectional
Quantification	Frequency (Freq)	s, min, h, d, w, mth, y	Typical frequency of communication per single link (order of magnitude, e.g. s = several times per second). (s = second, min = minute, h = hour, d = day, w = week, mth = month, y = year)
	Latency (Lat)	s, min, h, d	The order of magnitude of the acceptable latency of the communication (s = second, min = minute, h = hour, d = day).
	Bandwidth (Bwth)	B/s, KB/s, MB/s	Order of typical Bandwidth per second. Bit/s, Kbps, Mbps,
Security (Sec)		A(*x*), I, C	A(x) = Authentication required where x is/are the party which authenticates it's identity (possible values for x: Sr; Sk; Sr,Sk). E.g. A(Sr,Sk), source and sink needs to authenticate their identity before transmission.

Item	Sub-Item	Values	Description
			I = Data integrity. The integrity of data cannot be altered (by chance or on purpose) without the system noticing it. 'I' indicate that special algorithms such as signed message digests must be available. C = Confidentiality. Prevents a third party to overhear the communication. Encryption means must be available to protect data against disclosure. Symbols may be combined. E.g. A(Sr)/C: confidentiality must be insured and Source authenticated with strong mechanisms.
Availability (Avail)			Comments on the availability of this technology should be stated here, e.g. GSM can be used World-wide (roaming), TETRA and TETRAPOL are only regional / in some nations
Data	Type	Text, Audio, Video, Photo, Graphic, Animation, Multimedia, Raw data	Text: simple formatted text; Audio: simple audio record; Video: video record – no sound; Graphic: image which can be reconstructed out of a couple of data. Photo: still image; Animation: the image is not completely still some parts may be animated (moving icon for instance). Multimedia: the choice of the actual type is up to the application design. Raw data : formatted messages, queries, security data, etc. For physical transfer, a 'p' is put in front of the type (e.g. pText: physical transfer of Text). A combination of several type is possible (e.g. Audio/Video).
Comments			Definition of the session when needed, Comparable physical data flows, remarks on the way values were chosen, hypothesis, reason for not giving a value, etc.

6.2 Synthetic Table

The table comparing the characteristics of the different types of communications links is shown in the following pages.

Table 4 Synthetic table on telecommunication technologies linked with ITS

Technology	Name	Configuration				Quantification			Sec	Avail.	Data-Type	Comments
		Type	Con	Range	Dir.	Freq	Lat	Bwth				
Land based wireless telephony systems												
GSM			WL	Long					A			
	Voice	p-p			d	10 s for call set up	ms			World		The supplementary service "multi-party" enables the user to set up a conference call with several others
	Data (connection oriented)	p-p			d	10 s for call set up	ms			World		
	SMS	p-p, p-m			d					World		p-m only if direct connection to SMSC, roaming complicated (mainly caused by network providers)
	CB	p-b			u					some countries		
	GPRS	p-p, p-m, p-b			d					Coming		
	HSCSD	p-p			d					?		
DECT		p-p	WL	Med	d					local		
TETRA		p-p, p-m	WL	Long	d, sd					few countries		
TETRAPOL		p-p, p-m	WL	Long	d, sd					few countries		

Technology	Name	Configuration				Quantification			Sec	Avail.	Data-Type	Comments
		Type	Con	Range	Dir.	Freq	Lat	Bwth				
Satellite Systems												
GPS		p-b	WL	very long	u					world		system for exact positioning, exact time can be determined, main problem: American military system
IRIDIUM										world		financial problems.
Globalstar										world		chances to survive
Mass Broadcast Systems												
Radio			WL	Long					none			
	FM	p-b			u					Europe		
	DAB	p-b			u					few countries		
TV			WL	Long					none			
	analogue	p-b			u					world		
	DVB	p-b			u							
	Satellite	p-b			u					world		
Special Wireless Systems												
WLL-Technologies			WL	Short								
Beacon Systems		p-b	WL	Short	u				none			only special applications
Bluetooth			WL	very short								

Technology	Name	Configuration				Quantification			Sec	Avail.	Data-Type	Comments
		Type	Con	Range	Dir.	Freq	Lat	Bwth				
Fixed line systems												
Telephone			Wr									
	POTS	p-p			d		ms					
	ISDN	p-p			d		ms					
ADSL		p-p	Wr		d							
X.25 / X.31		p-p	Wr		d							
Frame Relay			Wr									
ATM			Wr		d							
Internet		p-p, p-m	Wr		d							
Ethernet			Wr		d							

7 Short introduction to the OSI model

The OSI (Open System Interconnection) Reference Model of network architecture and a suite of protocols (protocol stack) to implement it were developed by ISO in 1978 as a framework for international standards in heterogeneous computer network architecture. This standard is described in the ITU X.200 suite.

The architecture is split between seven layers, from lowest to highest: 1 physical layer, 2 datalink layer, 3 network layer, 4 transport layer, 5 session layer, 6 presentation layer, 7 application layer. For each layer it is possible to specify protocols and functions required for two nodes to communicate using the underlying network infrastructure (physical medium, switches, routers, bridges, multiplexers, intermediate nodes). Each layer uses the layer immediately below it and provides a service to the layer above.

Figure 8 OSI Model Layer Structure

Application Layer **7**	Programmes and applications	Mailer, Web Browser, etc.					
Presentation Layer **6**	Conversion system	ASN.1 ASCII translation ...					
Session Layer **5**	**N**etwork **O**perating **S**ystem	Netware IP/IPX - NetBios - DEC - TCP/IP					
Transport Layer **4**		NWIP, PCLAN, LanManager, DECNET, PC/TCP					
Network Layer **3**		ISO 8802.1 (IEEE 802.1)					
		ISO 8802.2 (IEEE 802.2)					
Access Layer **2**	**L**ogical **L**ink **C**ontrol	ISO 8802.3	ISO 8802.4	ISO 8802.5	-	ISO 9314	SDH
	Medium **A**ccess **C**ontrol	IEEE 802.3 CSMA/CD	IEEE 802.4 TOKEN BUS	IEEE802.5 TOKEN RING	IEEE 802.6 DQDB	ANSI X3T9.5 FDDI	
Physical Layer **1**							WDM

Detailed comments on each of the seven layers that make up the OSI Model are provided on the following page.

- Physical Layer. This is concerned with the transmission of signals. Choosing a solution at this level means choosing, for instance, fibber optics, copper wires, air and the equipment which actually create the signal.

- Data Link Layer. This is concerned with making the transmission reliable. Choosing a solution at this level means choosing algorithms to trigger the signal.
- Network Layer. This is concerned with routing data to the appropriate destination. Choosing a solution at this level means choosing a protocol which enable different sources and sinks to exchange the expected data and that only.
- Transport Layer. This is concerned with making the best use of the network on behalf of the entity which receives or sends the data. It is normally present only at both end of a communication channel. Choosing a protocol at this level means choosing a protocol which enables or not to multiplex data, to prioritise data, to control the flow, to detect/correct errors, to open or close a communication.
- Session Layer. This is concerned with providing means to establish a dialogue: open or close a conversation, synchronise exchanges etc. Choosing a protocol at this level means choosing how dialogues will be managed.
- Presentation Layer. This is concerned with formatting the information that applications are exchanging. Choosing a protocol at this level means choosing which common dictionary or message format will be used by both end. It may also mean to choose a protocol which describes in real time the exchanged data. (This is more or less the language issue described below).
- Application Layer. This is the highest layer were the information is processed before being handed over to lower layers for transmission, when not stored or handed to users.

In practice, this theoretical model is not followed by industrial telecommunication companies. *De facto*, protocols often cover several layers. For instance the ATM protocol covers part of the second, the third, and of the fourth layer. In its own manner GSM covers all layers. Additionally, they are not fully independent. Most of the time it is not possible to use any physical layer with a given protocol. For instance ATM was first designed for fibber optic networks.

8 References

(a) European ITS User Needs Deliverable Document, Issue 1, May 2000.

(b) European ITS Framework Architecture Functional Architecture Deliverable Document (D 3.1), Issue 1, August 2000.

(c) European ITS Framework Architecture Physical Architecture Deliverable Document (D 3.2), Issue 1, August 2000.

A copy of any of the above Documents can be found on the European ITS Framework Architecture CD-ROM, or at "http://www.trentel.org/transport/frame1.htm" by selecting "Deployment Information" and then "System Architecture".