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UTRAN Transmission Infrastructure Planning and Optimisation

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17.1 INTRODUCTION

The fundamental changes between 2G to 3G transmission networks and their resulting risks and chances are often overlooked in the first phase of a 3G network deployment. Most of the current mobile network operators originate from a classic telecom background and had only little experience and affinity to implement direct IP-vendors during the initial 3G rollout, which was often before 2004. On the other side, nearly all IP- and ADSL-operators do not comply with the same level of quality, which classic telecoms were used to. The reason is based on the different type of services. While data services, such as download, e-mails or surfing, can easily tolerate end-to-end outages of ten or more seconds, this is impossible for the performance of voice services, which has for a long time been the core product of mobile operators. 3G experiences a merge of both main service categories into one access network, which implies two main requirements for an access transmission network:

1. Quality to transport conversational services like voice and video telephony with the same availability standards such as a 2G telecom provider.
2. Cost efficiency to compete with IP-companies in offering high bandwidth services to customers.

Currently, most of the revenues are still being generated by voice services, which means that high availability standards still have to be ensured. With data services emerging, this strategy will change, particularly if a significant portion of them was not conversational or streaming. Furthermore, the option of throughput asymmetries in forward and reverse traffic lead to new challenges, since Plesiochronous Digital Hierarchy (PDH) and Synchronous Digital Hierarchy (SDH) transmission systems show here significant constraints.

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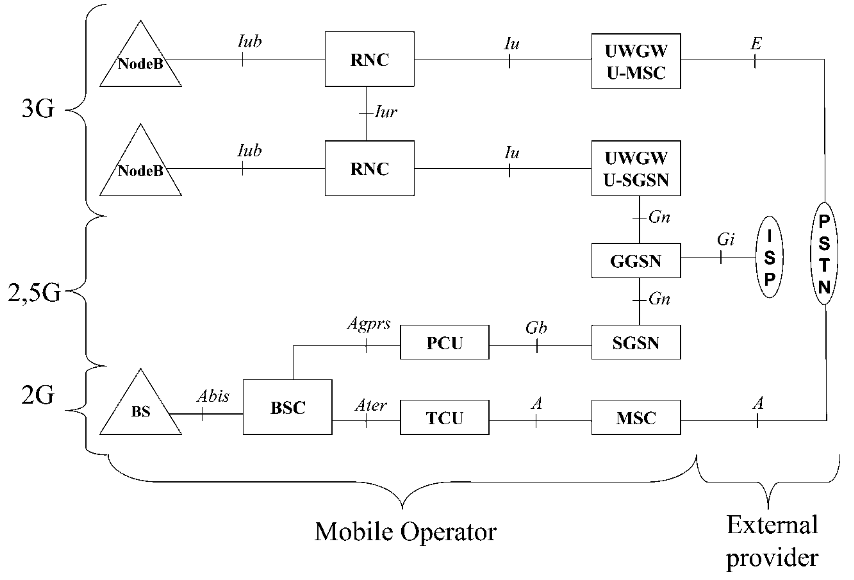


Figure 17.1 UTRAN overview.

17.1.1 SHORT UTRAN OVERVIEW

The UTRAN transport network has been standardised by the UMTS forum, 3GPP and 3GPPiP. Figure 17.1, derived from [1], [2] and [3], shows the UTRAN network nodes and interfaces and gives an example of the integration of 2G and 2.5G network nodes into the UTRAN access transmission structure.

The abbreviations can occasionally differ throughout the vendors, but the principles are similar: While 2G transmission networks are designed to transport all mobile-originated payload back to the MSC or SGSN this does not necessarily apply to 3G transport networks. Compared to a 2G BSC, a 3G RNC can execute the mobility management as well as the routing of traffic within an access network without routing it via an UMTS Wireless Gateway (UWGW), and UMTS Main Switch controller (U-MSC) or an UMTS Serving GPRS Server Node (U-SGSN). If call control features like authentication, user registration (HLR, VLR) and billing are implemented on RNC-level, then this setup allows a complete separation between the connectivity plane and the control plane.

17.1.2 REQUIREMENTS FOR UTRAN TRANSMISSION INFRASTRUCTURE

All mobile operators currently experience the paradigm change of a complete circuit switched network topology – mostly based on PDH and SDH-physicals – to a packet-oriented topology like Ethernet/IP routing. Since the very most of the mobile operators have implemented their transmission infrastructure based on circuit switched topologies, most of them use packet emulation technologies to transport packet data, in particular Frame Relay for 2.5G and ATM AAL2 or AAL5 for 3G data traffic. Compared to direct IP router technologies like Ethernet these emulators produce a significant transport overhead [4].

This has economical consequences, when high user data rates have to be transported. On the other hand, packet technology offer a great variety of routing options allowing a more flexible and thus economical utilisation and scalability of physical infrastructure. A variety of wireless transport systems are available in several markets. Beside conventional PDH and SDH microwave links as well as fibres, new wireless systems like

- PMP-systems based on ATM or IP cross connects (see Section 17.4.3),
- WiMAX systems (see Section 17.4.4), and
- Wireless Gateways from various vendors

exist now on the market. If allowed by the national regulator, this offers new, better performance and cost-efficient opportunities to cope with the requirements.

17.1.2.1 Technical Requirements

Compared to circuit switched traffic, a guarantee of service quality for dedicated sites or services is difficult, because neither ATM nor IP evaluates its payload on transport level (layer 2). The ATM Adaptation layers (AAL1, 2, 3/4 and 5) and service classes (CBR, VBR, UBR, ABR) can prioritise services on the session level (layer 3), provided that the transport of such services was enabled end-to-end on dedicated Virtual Channels (VC). If not, the only option is the encapsulation of session bits on OSI-layer 4 or higher to ensure service quality [5]. Beside the service quality and network availability, further requirements have to be met:

- Accessibility and interoperability with third-party carriers, particularly ISPs. Particularly the establishment of virtual home environments incorporating high data rate services via two or more carriers requires optimised UNIs (User Network Interfaces) and quality enablers.
- Insurance of traffic contracts and policing, if required by third-party carriers, corporate customers or companies.
- Interoperability with 2G and 2.5G user equipment as well as with all 3G user equipment types defined by IMT-2000.
- Guarantee of maximum delay standards for interactive and streaming classes. Most of the vendors tolerate a net transmission delay of 10 ms between Node B and RNC depending on the delay budgets of its components. Since ATM introduces significant packetisation and transport delays, and IP v4 can produce remarkable defragmentation delays, the limitations can be severe.
- Assurance of ITU synchronisation standards, particularly [6–8]. While the synchronisation of all 2G and UTRAN network elements via SDH and PDH can usually be established, there are limitations to a number of IP routers.
- Insurance of minimised transmission overheads as well as retransmission on packet transport. This includes the establishment of channels that enable bandwidth-on-demand. This aspect has become a critical issue for the initial 3G transmission network products. Some UTRAN-vendors keep code capacity allocated during a session even when no user traffic is present. This leads to a significant transmission overhead, particularly during web page changes or interactive gaming. Other vendors re-establish instead a session when new data is transmitted. This leads to significant delay or even drops on a congested network, when for example the CAC (Call Admission Control) delays a PVC (Permanent Virtual Circuit) or SPVC (Semi-Permanent Virtual Circuit) re-establishment.

With the growing complexity of services, user equipment and data throughput the requirements to meet technical KPIs will rise drastically. However, the economical objectives will become increasingly important in the future as well.

17.1.2.2 Economical Requirements

There are many requirements for 3G networks that are not driven by technical but by economical needs. Mobile operators will enter a direct and stiff competition with Internet service providers. Presumably 50 % of the mobile operators' revenues within the next 5 years will not be generated by airtime but by portal content. ASPs will be in the position to choose and prioritise between mobile and fixed carriers. However, capacity roadmaps and price erosion of IP-vendors occur significantly faster than that of telecom vendors, putting IP-carriers in an economical advantage. The recent growth of WLAN hotspot technology is just one example for the potential of wireless IP carriers.

Particularly the economic circumstances will be the main driver for the backhaul network strategy. Unlike the radio access network, these strategies cannot be standardised but will be unique to each operator. This has the following reasons:

- The access transmission network has no direct customer visibility. For this reason it does not have to be designed at all in a plug-and-play manner like air interface user equipments. In fact, each of the ATM vendors differ in the detailed form in which they have configured VCs, Interfaces or PNNI (Private Network-to-Network Interface).
- Each country has a different leased line carrier structure offering various products and prices. With 3G nodes being able to operate layer-1 products like PDH and SDH as well as higher layer-products like Frame Relay, ATM or Ethernet, the options are tremendous.

Beside the economical constraints and opportunities defined by the market and the equipment vendors, some more requirements defined by the operator's product marketing have to be met:

- An entirely different billing and tariff structure. Billing can be applied on time, volume, number of sessions or interactions (like number of web pages) as well as on various type of content. This differs significantly from 2G applications where one call data record retrieved from the MSC is sufficient for most service applications.
- Enhanced options for resilience and dynamic routing. Conventional static $1+0$ protection of circuit switched connections is cost intensive compared to a dynamic routed meshed network.
- The ability to route sessions via the core network edges (like the Iur) instead via the switch. Since all UTRAN elements including Node B are equipped with ATM or IP functionalities the packetisation, transport and routing of user payload of and to virtually each network node is a scenario. And if traffic does not have to be routed to the switch, then the backhaul network can be relieved.

The development of decentralised routing and billing in the mobile access networks is comparable to the migration of IT-mainframe systems to Client/Server architectures. The result is a better scalability of the required network infrastructure and cost.

17.2 PROTOCOL SOLUTIONS FOR UTRAN TRANSMISSION INFRASTRUCTURE

Currently, the 3G mobile transmission protocol solutions are almost all based on ATM. End-to-End-IP, if configured, is very often operated on ATM AAL5 or on Frame Relay. Section 17.2.3 will describe the way to IP-network protocol solutions, which is seen as the path to the future mobile network architectures.

17.2.1 MAIN CONSIDERATIONS FOR ATM LAYER PROTOCOLS IN CURRENT 3G NETWORKS

ATM was developed by the ITU Study Group XVIII Standardization Sector as a Wideband-ISDN-standard in the late 1980s [9]. The fact that it became by far the most used layer-2 technology by 3G-operators was based on its ability to establish stringent quality standards for circuit switched as well as for interactive and background packet services. ATM Packets have been standardised as 53-byte packets, with 7 bytes required for the header [10] as shown in Figure 17.2.

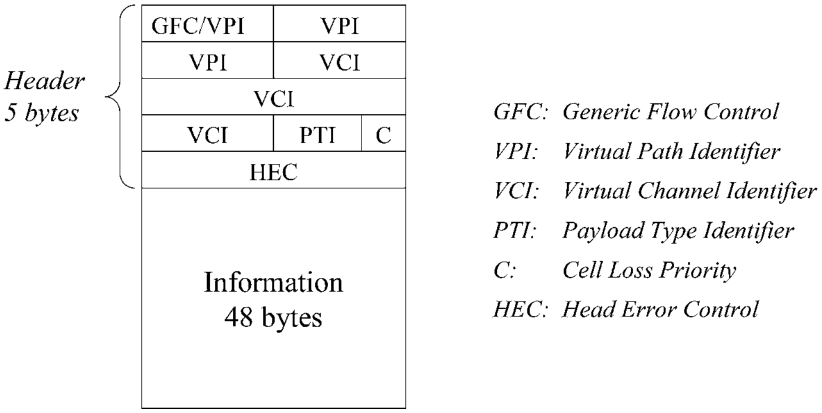


Figure 17.2 ATM cell structure.

Generic Flow Control (GFC) is only used in UNI, and supports the user configuration. The Virtual Path Identifier (VPI) field contains the address of the virtual path and incorporates usually several virtual channels. The channels are addressed in the Virtual Channel Identifier (VCI). The Payload Type Identifier (PTI) describes the type of payload. Payload can be distinguished between user and network data. The content of the CLP-field (Cell Loss Priority) triggers whether a cell will be dropped in case of congestion. Cells with CLP = 1 will be dropped earlier than cells with CLP = 0. The Head Error Control (HEC) field is required for error control and correction of the header data. With HEC, an ATM receive node can synchronise to the ATM cell offset. For error detection, a CRC-mechanism is used, which is based on the division of the header field by the polynomial $x^8 + x^2 + x + 1$.

Similar to all other interfaces, Iub traffic can be divided into traffic dedicated to user services (like Circuit Switched and Packed Switched Traffic), traffic dedicated to Node B signalling like C-NBAP (Common Node B Application Protocol) and D-NBAP (Dedicated Node B Application Protocol) and traffic dedicated for operation and maintenance (OAM). In ATM, each traffic type can be assigned to a VC (Virtual Channel) on a dedicated VCI. The VC is then mapped on a VP (Virtual Path). This assignment is done in the ATM-CC-Unit of the Node B or the RNC. VCs can be configured as permanent connections on a fixed route (PVC) or as switched connection (SPVC). SPVCs appear as permanent connections if seen from the ingress and egress port of establishment. Between the ingress and egress port, the route can be established dynamically depending on traffic load and congestion within the network. In general, it is desirable to have the option of a complete dynamic routing between the Node B and the RNC. However, if the ATM vendor differs from the 3G-vendor, the establishment of an end-to-end dynamic network is often impossible due to missing vendor interoperability to establish a PNNI. In such cases, some sections have to be designed as PVC originating on UNI.

To handle load and overbooking on an ATM trunk during connection establishment, the first measure is the configuration of a suitable CAC to each PVC or SPVC. During a session, the efficient utilisation of the ATM trunk can be ensured by the design of a suitable 'transmit-corridor' by applying a peak, minimum or sustainable cell rate. Example: An E1-Trunk consists out of 30 transmission channels (64 kbps) for payload plus 2 for transmission management. The peak information rate (PIR) is then 1920 kbps, which corresponds to a peak cell rate of PIR (in Byte/s) /53 byte, which means 4.528 cells/s. When the E1 resource shall be used for up to five Node Bs (overbooking), the minimum cell rate – which has to be guaranteed by the ATM-CC – is then 9.05 cells/s.

The prioritisation of PVC towards other PVC can be controlled by the application of a traffic class [11]. The traffic classes defined by ATM forum [10] are:

- Constant Bit Rate (CBR), which is real circuit emulation. In this category, the ATM network receives a continuous stream of bits. It usually implies a very low delay and very low delay variation.
- Real-Time Variable Bit Rate (RT-VBR). This service class has very tight bounds on delay but might not have very tight bounds on cell loss. There are certain kinds of traffic such that if the delay gets too large it might not deliver it at all.
- Non-Real-Time Variable Bit Rate (NRT-VBR) is the complement of RT-VBR. This class puts a low priority in the delay but focuses into not losing cells instead. E-mail service is an example of this type of traffic.
- Unspecified Bit Rate (UBR) is kind of "best effort" method, since UBR has no guarantees.
- Available Bit Rate (ABR) involves flow control. The goal here is to have a very low cell loss within the network.

With this set of options, the dimensioning of a well-performing and cost-efficient transport network is large. However, the first UTRAN vendor releases show severe limitations with regard to the described ATM functionalities, particularly a limited number of traffic classes, which cannot be translated from one ATM vendor to another. Additionally, the separate end-to-end transmission of different services (conversational, interactive, etc.) has not yet been established by several vendors or vendor interoperability, which means that most of the traffic has to be transported in one or two VCs (like one for circuit switched and one for packet switched traffic), and not in VCs dedicated to a specific access bearer service.

Beside the design limitations, several other constraints with regard to performance retrieval and optimisation exist. This will be described in the next sections.

17.2.1.1 ATM Adaptation Layers for Different 3G Bearer Services

The task for the ATM Adaptation Layer (AAL) is the adaptation of the payload data from upper layers to the format of the ATM cell. The adaptation happens in relation to the required services. Furthermore, it reassembles the payload stream at the ATM egress node and equalises cell delay variation. To satisfy the various requirements of the different services, four layer types have been created, namely AAL1, AAL2, AAL3/4 and AAL5.

All AAL are divided into the CS (Convergence Sub-layer) and SAR (Segmentation and Reassembly Sub-layer) [9]. The SAR assembles the upper layer data into segments optimised for the layer type size. The CS performs error correction, re-synchronisation as well as error checks.

AAL1. This standardised protocol is used for the transport of time critical applications and conversational services with a constant bit rate such as voice or video telephony. In addition, it is used for the emulation of circuits like E1, T1 or DS0.

The AAL1-SAR requires one of the 48 payload bytes therefore reducing the user payload to 47. The SAR header byte consists of a 4-bit sequence number and a 4-bit SNP, which generates a CRC-3 checksum. The AAL1-SAR adds error detection by calculating 4 bytes out of 124 data bytes and adding them into the cell stream. The entire 128-byte load is then reassembled as depicted in the Figure 17.3 [12]. This means that at least 128 cells have to be buffered before reassembling and transmission can begin [10].

AAL2. This type has a particular use in mobile applications since it is able to transport time critical services that incorporate a variable bit rate. AAL2, as defined in ITU-T-I366, supports CBR, RT-VBR and NRT-VBR. The principle of the AAL2 payload is illustrated in the Figure 17.4 [13]:

Each VC is assigned a VCI, which is stored in the CID-byte (Channel-ID-Byte, which indicates the Virtual Channel) after the ATM header. With VCI-Nr 0-7 reserved, it means that 248

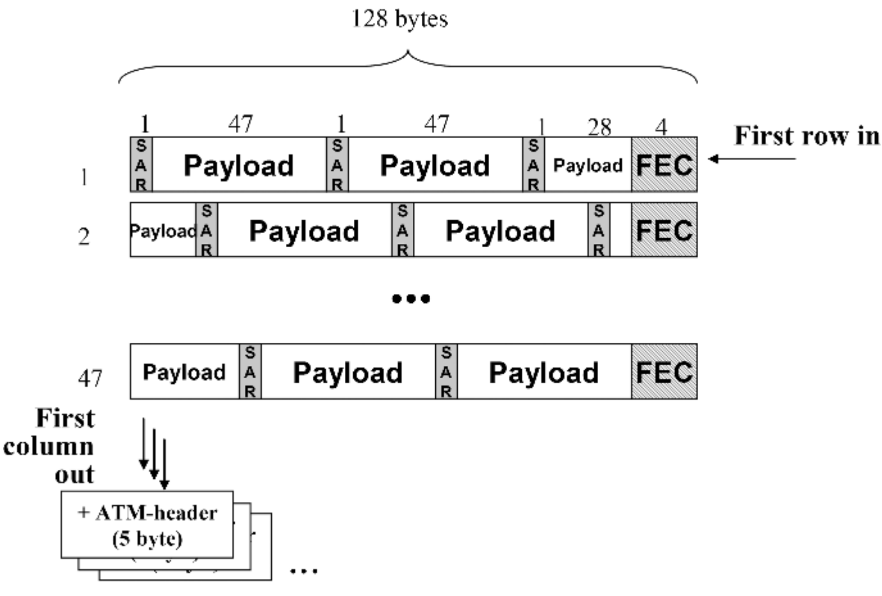


Figure 17.3 AAL1 structure.

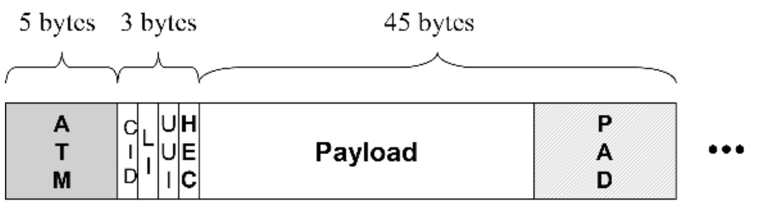


Figure 17.4 AAL2 structure.

VCI can be mapped to one VP. Two more bytes of the ATM cell are needed for further AAL2 functions:

- LI (Length Indicator): indicates the length of the payload (0–45 bytes). If the payload is smaller than 45 bytes, the rest of the cell will be filled with padding octets;
- UUI: (User-to-User-Indication). This indicator provides an information link between CPS (Common Part Sublayer) and SSCS (Service Specific Convergence Sublayer);
- HEC: Header Error Control (see FER in AAL1 description).

As a difference to AAL1, which supports only CBR, the maximum delay and minimum throughput of a channel can be configured by setting a MCR and a SCR as well as limits for the cell transfer delay and variation.

AAL3/4. The main function of the AAL3/4 type is the adaptation of connection-oriented and connectionless data transfer to the ATM cell format. Its main area of application is the connection of LANs and ATM transmission.

In AAL 3/4, the protocol first inserts error-checking functions before and after the original data. Then the information is segmented into 44-byte packets. The cell payload includes two bytes of header and two bytes of trailer so this whole construct is exactly 48 bytes. There is a CRC check on each cell to

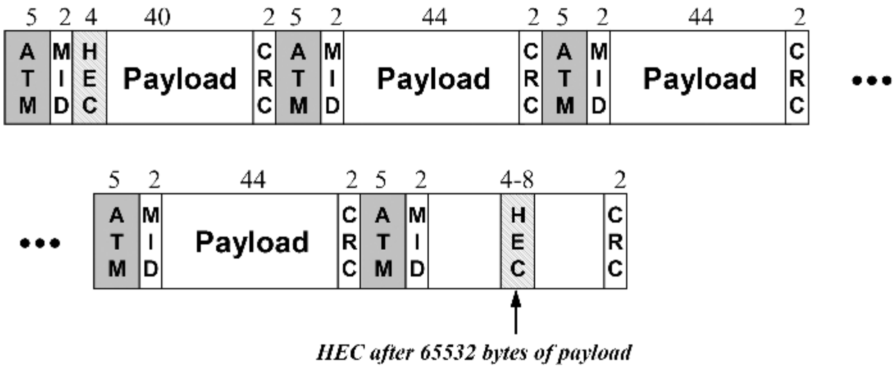


Figure 17.5 AAL3/4 structure.

check for bit errors as well as Message Identifier (MID). The MID allows multiplexing and interleaving of large packets into a single virtual channel. This is useful in a context where the cost of a connection is very expensive since it would help to guarantee high utilisation of that connection [14] (Figure 17.5).

Cyclic redundancy check is performed in each ATM cell, while HEC is performed after 65532 bytes of payload.

AAL5. The AAL5 type has been created for special requirements of packet-oriented applications. AAL5 is a downgraded version of AAL3/4 but with a lower overhead. Multiple conversations may not be interleaved in a given connection. Here the CRC is appended to the end and the padding is such that this whole construct is exactly an integral number of 48-byte chunks. This fits exactly into an integral number of cells, so the construct is broken up into 48-byte packets and put into cells [15]. Figure 17.6 shows AAL5 structure.

Table 17.1 gives a summary of all AAL and traffic quality options available, and assigns them to the applications and the bearer classes defined by the ATM Forum [16].

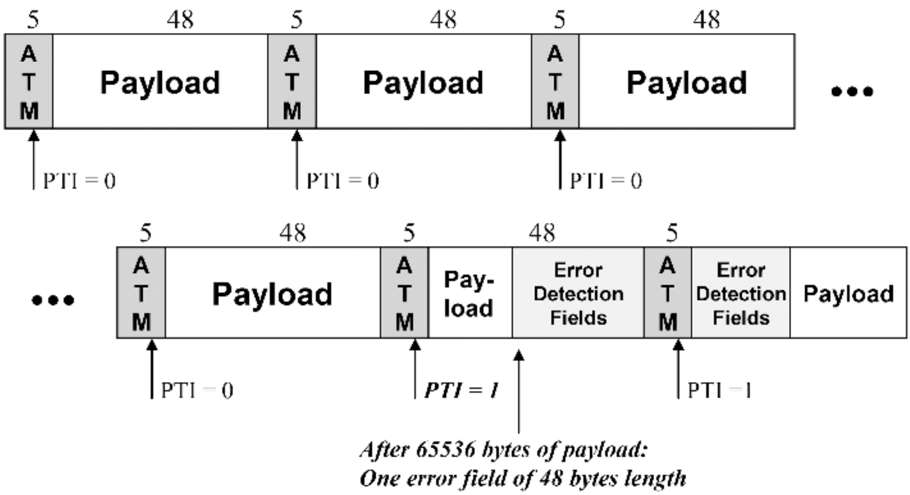


Figure 17.6 AAL5 structure.

Table 17.1 Summary of AAL types and qualities.

Service	CBR	rt-CBR	Nrt-VBR Connection oriented	Nrt-VBR Non-connection oriented	UBR	ABR
Bearer Class	Class A	Class B	Class C	Class D	Class X	Class Y
Applications	Voice and Clear Channel	Packet, video and voice	DATA			
Connection Mode	Connection oriented			Connection less	Connection oriented	
Bit rate	Constant	Variable				
Timing	Required		Not required			
Services	Private Line	None	Frame Relay	SMDS	Raw Cell	
AAL type	1	2	3/4 and 5	3/4	Any	3/4 and 5

17.2.1.2 ATM Key Performance Indicators

ITU has defined a set of standards for OSI layer 1, 2 and 3-performance retrieval on ATM and IP-systems, which are illustrated in Figure 17.7.

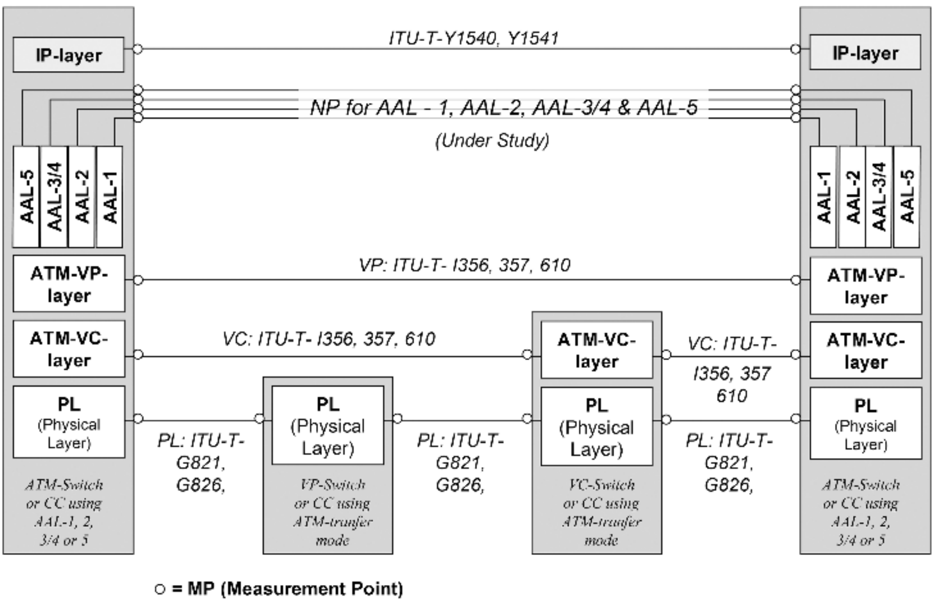


Figure 17.7 ITU Performance Measurement Regulations for physical layer, ATM and IP.

The main specifications for ATM performance are defined in [17–19]. The principle is comparable to the definition of unavailable and errored seconds, which are for example defined for SDH in [20]. For ATM systems ITU substitutes seconds by 53-byte ATM cells. Between two MPs (Measurement Points) six cell events are defined as in Table 17.2.

The second main criterion beside Cell Error is the transmission delay as well as the delay variation of the ATM cell stream [21]. Cell Transfer Delay (CTD) and Cell Delay Variation (CDV) are illustrated in Figure 17.8.

The mean Cell Transfer Delay (CTD_{mean}), 1-Point Cell Delay Variation at MP2 ($CDV_{\text{Cell}}^{\text{1Point}}$), and 2-point Cell Delay Variation between MP1-MP2 ($CDV_{\text{Cell}}^{\text{2Point}}$) are defined as follows:

$$CTD_{\text{mean}} = \frac{1}{n} \cdot \sum_{i=1}^{n} \Delta t_{\text{i}},$$

$$CDV_{\text{Cell}}^{\text{1Point}} = t_{\text{cell}} - D_{\text{ref}},$$

$$CDV_{\text{Cell}}^{\text{2Point}} = \Delta t_{\text{cell}} - D_{12},$$

where:

D_{ref} – reference point of time that an ATM cell should reach the point MP2,
t_{cell} – point of time that the specific ATM cell reaches Time reaches MP2,
D_{12} – reference delay for an ATM cell between MP1 and MP2,
Δt_{cell} – delay that the specific ATM cell has required between MP1 and MP2.

ITU gives recommendations for KPI thresholds, but each operator will set its own values depending on its hardware, network or Leased Line infrastructure, products and Service Level Agreements (SLA).

17.2.1.3 New Challenges and Opportunities Using ATM Transmission

A real challenge for ATM transmission is the correct and quick retrieval of its end-to-end performance. Also, the detection of traffic congestion spots in meshed networks, particularly when overbooking, IMA (Inverse Multiplexing on ATM) and PNNI are configured, can become a real challenge. Additionally, the KPIs for ATM performance have only been ITU-specified for PVC, but not yet for SPVC. The tools for optimisation are illustrated in Section 17.2.1.5.

In the classic 2G-world, fault detection and optimisation of e.g. a GSM-circuit is commonly handled on a section-by-section basis. This is not applicable for a PNNI-'cloud', because a SPVC-route cannot be mapped at all to a physical transmission system. For this reason, it is required to ensure performance and optimisation on both the physical systems and the logical network connections [9]. We shall proceed with some examples.

Usage of IMA. IMA usually groups several physical ports (e.g. E1, T1, E3) to a virtual (logical) port. This accelerates ATM packetisation and reduces the complexity to route a VC, because it is then mapped exclusively to this logical port. Likewise, all retrieval of utilisation or errored cell data is based on this virtual port. A malfunction of a port in an IMA-group is then less easy detectable, particularly when the malfunction is not generating an alarm [10].

Usage of dynamic routing in a PNNI-cloud. Most of the operators configure PNNI, because it increases the bandwidth efficiency, which reduces cost. This, however, means that an SPVC can be routed via any physical connection and route in this cloud. This requires that certain KPIs and components to be constantly monitored and optimised:

Table 17.2 Events defined by ITU-T I356 and I357.

Event	Description	KPI	
STCO (Succesful Transferred Cell Outcome)	The cell has been transferred without error		
ECO (Errored Cell Outcome)	The cell header or payload has been transferred in error	CER (Cell Error Rate)	$\text{CER} = \dfrac{\text{EC (Errored Cells)}}{\text{STC (Succesful Transferred Cells)}}$
TCO (Tagged Cell Outcome)	The cell has been tagged due to a transmit delay	CLR (Cell Loss Ratio)	$\begin{cases} \text{High Priority cells } \text{CLR}_0 = \dfrac{\text{LC (Lost Cells)} + \text{TC (Tagged Cells)}}{\text{TTC}_0\text{(Total Transferred Cells w.CLP} = 0)} \\ \text{All cells} \quad \text{CLR}_{0+1} = \dfrac{\text{LC (Lost Cells)}}{\text{STC}_{0+1}\text{(Total Transferred Cells)}} \\ \text{Low Priority cells } \text{CLR}_1 = \dfrac{\text{LC (Lost Cells)}}{\text{TTC}_1\text{(Total Transferred Cells w. CLP} = 1)} \end{cases}$
LCO (Lost Cell Outcome)	The cell has been lost		
MCO (Misinserted Cell Outcome)	The cell has been misinserted in the received ATM cellstream	CMR (Cell Misinsertion Rate)	$\text{CMR} = \dfrac{\text{MC (Misinserted Cells)}}{\Delta T\text{(Time Interval)}}$
SECBO (Severely Errored Cell Block Outcome)	An entire block of ATM cells has been received in error	SECBR (Severely Error Cell Block Ratio)	$\text{SECBR} = \dfrac{\text{ECB (Errored Cell Blocks)}}{\text{TTCB (Total Transferred Cell Blocks)}}$

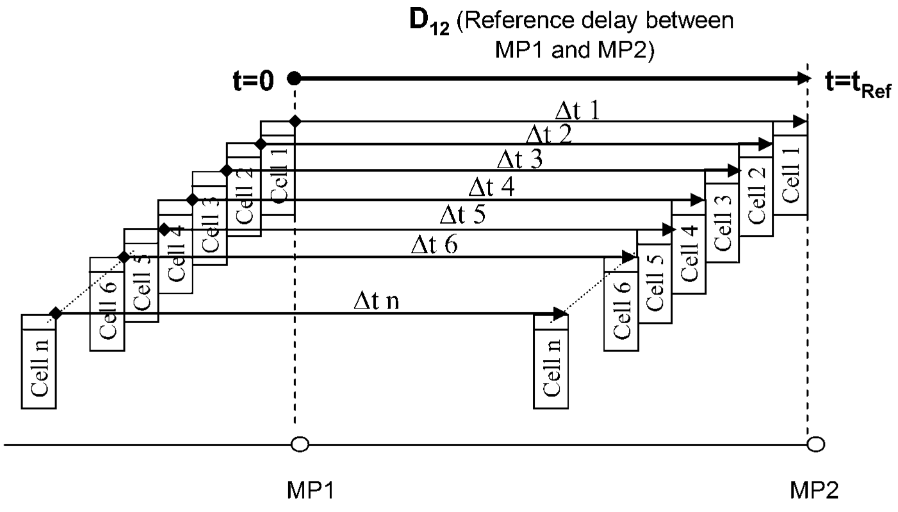


Figure 17.8 ATM transfer delay.

1. The load and cell performance of all physical ports and interfaces in the cloud.
2. The load and cell performance of the monitored connection.
3. The average length of the established SPVC-route, which determines the average cell delay.
4. The average length of an SPVC-reestablishment.

Particularly (3) and (4) can drastically increase with the load and complexity of the ATM cloud. This can have fatal consequences. If a SPVC is re-established due to congestion or physical unavailability, it does not commonly generate an alarm to the UNI. Depending on the service transported on the SPVC, this leads to call or session drops or even service resets. This is of particular concern in conversational services [9].

Overbooking. ATM allows overbooking of physical trunks with PVC or SPVC peak cell rates. This means that if a PVC has been established but the traffic volume decreases, this volume can be filled by traffic of other PVC. As a consequence, PVC can transmit a higher volume than its configured Peak Cell Rate (PCR) if the physical environment allows it, and if traffic policing and shaping are disabled. It should always be remembered that PCR is the criteria for the Connection Admission Control during PVC establishment, but not during its entire transmission [9].

Another new challenge and opportunity is the emulation of PDH or SDH circuits on OSI-Layer 3 (Circuit Emulation Services, CES). The design for CES transport will be described in Section 17.2.1.6.

17.2.1.4 Dimensioning Link Load and Occupancy in ATM Networks Using PNNI

Dimensioning the load of ATM networks carrying mobile application traffic depends on the service mix. A safe approach to create a rule of thumb is to split all services into two main services:

1. Conversational services like voice or video telephony as well as data services that require CBR quality.
2. All remaining services, which are more background based or interactive.

For both type of services, 2G and 2.5G have developed dimensioning methods that also can be applied on a service mixed port or connection: conversational services have been successfully dimensioned by the Erlang B formula for a long time. If several service bit rates were applied, the usage of its multirate derivation is applicable [22].

For background and interactive services, the dimensioning experience of mobile data networks, particularly WAP and GPRS, can be applied. Most vendors here recommend a maximum GPRS-Bearer (Gb) load (see Figure 17.9) of not more then 80 % of the accommodating Gb-capacity, because of the burstiness of data traffic. Taking into account the session and link layer overhead (DLCI-framing, PDP-context, etc.), this means that the user payload should not exceed more than 55 % of the Gb-capacity. However, it has to be considered that the real overhead for interactive services is significantly higher because the radio channel bearer set-up is not immediate therefore decreasing its throughput after a user interaction. For this reason, a safe design threshold is the dimensioning of no more than 70 % of the entire connection or interface capacity, if the services are entirely non-conversational [23].

With the further consideration that conversational services have to be prioritised and the ATM overhead for CBR and AAL1 can be considered as a stable 17.5 % (see Section 17.2.1.1), the dimensioning method for PVC without overbooking can be performed as depicted in Figure 17.9; the dimensioning of required bandwidth for data services by 120 % / 70 % is based on the minimum ATM AAL2 overhead plus the 70 % threshold described earlier.

If overbooking and dynamic routing is introduced, the additional risk of congestion has to be considered, the dimensioning threshold depends heavily on the overbooking factor. It is recommended that the engineering limit for a SPVC five times overbooked ATM trunk or ring should be reduced to 50 % instead of 70 %.

The described dimensioning approach is of particular interest if the load and the ATM performance cannot be retrieved on an end-to-end basis or the load needs to be simulated by software tools. For more in-depth evaluation and optimisation, live network tools are required, which are described in Section 17.2.1.5.

17.2.1.5 Optimisation of ATM Network Parameters

Similar to the RF-Interface, the optimisation of ATM networks is performed in two steps [22,24]:

1. Optimisation of nodes.
2. Optimisation of clusters.

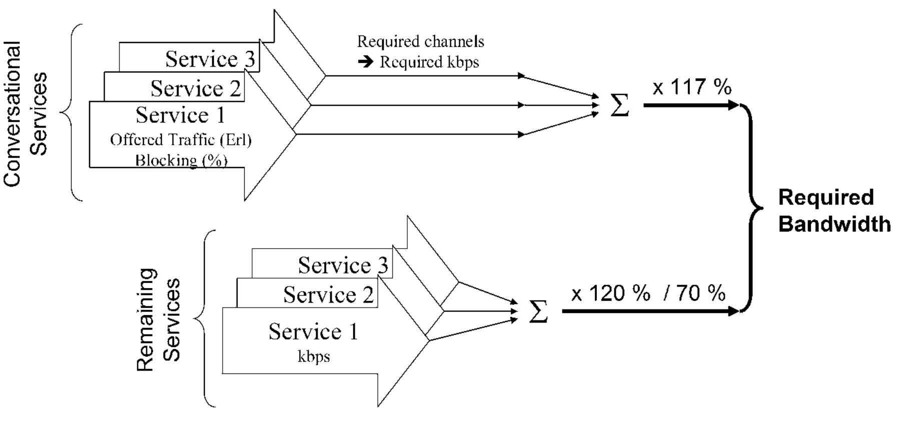


Figure 17.9 ATM dimensioning method.

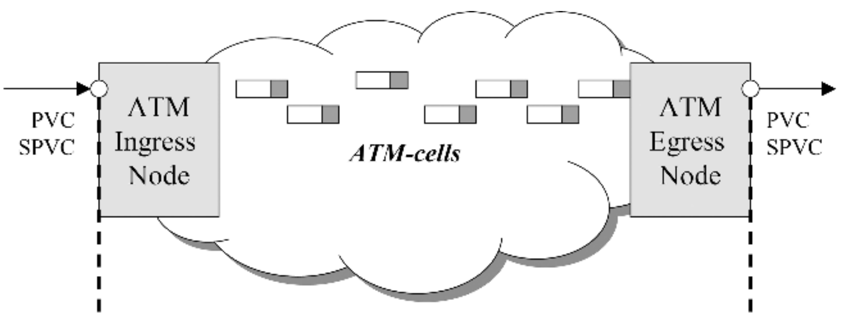
In step 1, all physical and logical connections have to be checked. The checklist contains issues like:

- Synchronisation: Are all nodes properly synchronised? When did the transmission systems undergo a 24-h stability check for its last time? How is its CRC-4, LSS (Loss of Synchronisation Signal) or AIS-record (Alarm Indication Signal)? Has it been considered that only AAL1 and AAL2 connections require and supply timing? Etc.
- Service classes: Are PVC-connections within an end-to-end-route configured with the same service class (CBR, VBR, ABR, UBR). Do all ATM vendors and components in the network support and interoperate these classes? Have they been applied with the desired AAL types? Etc.
- Physicals: Are the physicals (e.g. the Node Bs) evenly distributed to the ATM ports? Are the ports, the cards and the connections properly protected? Etc.
- Throughput: Are the PCR, MCR and SCR configured correctly? Are they plausible considering the expected service mix and throughput? Does the combination leave sufficient throughput for UBR and ABR-traffic? Etc.
- Cell loss: Are the buffer sizes and the cell delay tolerances implemented as recommended by ATM Forum or the component vendor? How is traffic policing and shaping implemented on this component, this port or this VC? Etc.

In step 2, the end-to-end performance has to be established by an entire ring or mesh optimisation [25]. The questions for this session can be quite different:

- What average Cell Transfer Delay (CTD) and Cell Delay Variation (CDV) can be retrieved from the system? What or how many ATM nodes are in the route on average? Etc.
- How high is the utilisation of the ports? Can clusters or nodes be identified that are over- or under-utilised? Etc.
- Are Peer Groups configured in a way that meshes or rings are divided? How much load is on the Peer Group lead component? Etc.
- Do the SPVC or PVC contain significant amounts of tagged, errored and lost cells? At which component do they occur on the way? Are they retrieved properly? Etc.

It can generally be stated that, the more thoroughly step 1 is performed, step 2 takes a much shorter time. Step 2 requires the implementation of specific tools that are specified e.g. in ITU T-I610. Some ATM vendors, however, have not yet implemented tools like this and almost all UTRAN-vendors do not offer I610 Performance retrieval at all. Instead, they retrieve performance like CLR as depicted in the scheme in Figure 17.10.



$$CLR = \frac{\sum Cells\_Tx\_Ingress - \sum Cells\_Rx\_Egress}{\sum Cells\_Tx\_Ingress}$$

Figure 17.10 ATM Performance retrieval without I610 tools.

This scheme, however, does only work for utilisation, but not at all for error or loss performance.

Example: Take an E1-connection between Node B and RNC with an average CTD of 10 ms. Assume further that a management system is able to retrieve the valid cells transmitted and received every hour exactly synchronised with μs-accuracy. Even in this case, 55 of 19,660,800 ATM cells in an hour would be within the 10 ms route giving an inaccuracy of already $2.7 \cdot 10^{-6}$, making it impossible to establish KPI of $1 \cdot 10^{-6}$ resolution as defined by ITU.

In real life operations, it will not be able to synchronise retrieval even in one second accuracy, making any result retrieved on a two node basis completely unusable.

For this reason ITU-T-I610 defines the establishment of tools that add ATM cells containing accurate information on the amount of cells and a time stamp. The process is depicted in Figure 17.11.

The implementation of these tools allows an accurate retrieval of CLR, CMR and CER. In general, [19] defines four optimisation methods and tools:

1. Alarm Indication Signal/Remote Defect Indication (AIS/RDI): This tool monitors any indication of an upstream interruption at the ATM (or physical) layer and generates an alarm.
2. Loopbacks: This method works similar to an IP ‘ping’. It is useful for example when provisioning an ATM connection. Note that these ATM layer loop-backs are not to be confused with physical layer loop-backs such as may be created by manually inserting bridging cables at the remote end of a physical layer path.
3. Continuity Check (CC): This tool detects an interruption in an ATM connection (e.g. in an ATM switching matrix).
4. Performance Monitoring (PM): This tool detects and measures errored cells or lost/misinserted cells in an ATM connection.

The implementation of ITU-I610 is not cheap, making it not likely that all UTRAN vendors will implement it soon and thoroughly. For this reason, it is important that it is at least available at the core sections of the transmission network.

17.2.1.6 Exploitation of 3G-ATM Networks for 2G Voice and Data Networks

Several operators are in the position of having implemented separate backhaul networks for 2G and 3G services. This situation is an economical driver in the coming years to shift 2G and 2.5G traffic onto the ATM networks that have actually been implemented for the transport of 3G. One main reason

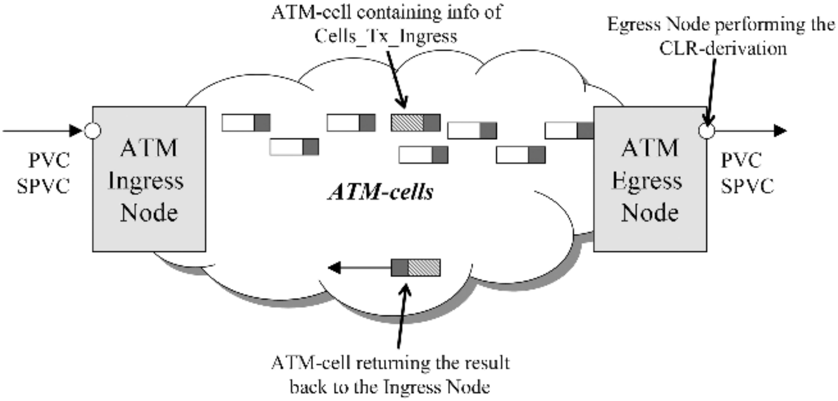


Figure 17.11 ATM performance retrieval using ITU-T-I610 tools.

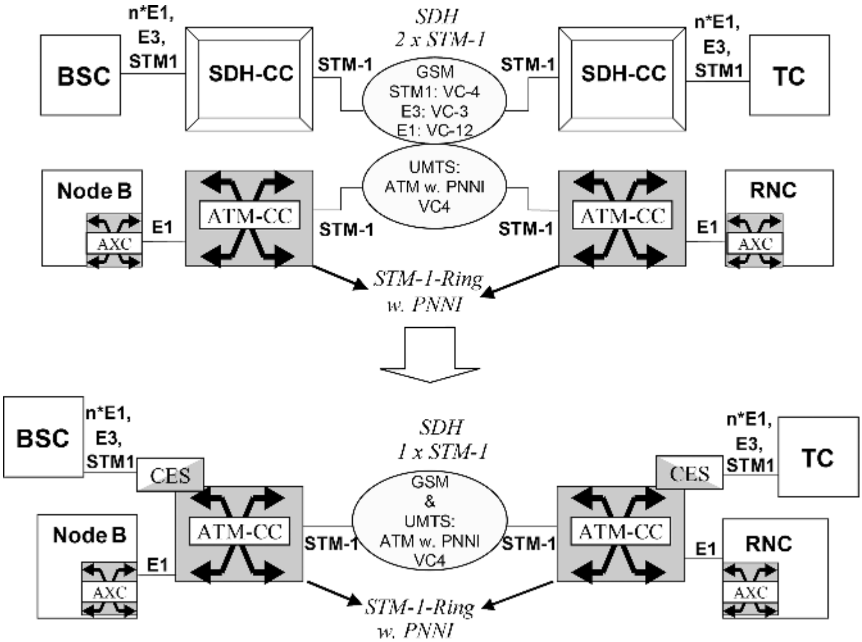


Figure 17.12 Circuit Emulation on ATM.

is a feature that has been standardised by the ATM Forum, namely CES (Circuit Emulated Services) as shown in Figure 17.12. CES emulates a physical E1, E3 or STM-1, converts the emulated payload to ATM, and then transports the signal via the ATM Network [24].

This has several economical advantages:

- Only one physical trunk has to be used for the transport of 2G and 3G traffic, which means no or later upgrades of microwaves, Leased Lines, fibres, switches or cross connects on growing traffic.
- The implementation of resilience for several 2G interfaces is easier and thus more cost-efficient using dynamic routing features like SPVC-reestablishment and PNNI instead of a $1+0$ protection.
- The complexity of the backhaul network is lower, which has a positive impact for maintenance, because ATM allows several QoS classes; the prioritisation of 2G, 2.5G or 3G traffic can be configured on demand.

According to the number of ATM specifications, a variety of configurations can be implemented on port level. Table 17.3 shows some options used for an E1 port as well as the impact on the packetisation delay (the delay to pack the bits on the PCM frames into one ATM VC) as well as the bandwidth saving [24].

In general, ATM generates an overhead of at least 17.5 % when CBR and AAL1 is used. Considering that an ATM frame on PDH takes 125 μs, this results in a minimum delay of 367 μs to pack and unpack 2048 bit into ATM cells. When reducing the payload by half – for example by transmitting 16 of 32 DS0 or compressing the 64 k DS0 to 32 k – the delay doubles, while the required bandwidth is reduced to 58.3 %. In the field some microseconds have to be added to the equipment. Secondly, every additional ATM hop generates an additional delay of 200 μs, if STM-1 transport is used. IMA will add 1.5 ms to the delay.

Actually, the main delay limit according to GSM-spec 3.3 is defined to be 180 ms round trip between the mobile and the transcoder unit. Considering the actual delay budgets of 2G network elements (for

Table 17.3 E1 Trunk types using CES.

CES type	Description	Packetisation delay (Ingress + output port)	Bandwidth compared to E1
Unstructured	Emulation and transmittal of the entire E1-trunk	≥367 μs	≥113 %
Structured and unchannelised	Emulation and transmittal of all 31 DS0 of an E1-Trunk, except Time Slot 0	≥378 μs	≥109 %
Structured and channelised	Emulation and transmittal of between 1 and 31 DS0 of an E1 Trunk	378 μs–11.75 ms	109 %–3.52 %
Structured and channelised and compressed	Emulation of 1 to 31 DS0, compression from 64 kbps to 32, 16 or 8 kbps, conversion and transmission over ATM	378 μs–46.0 ms	109 %–0.88 %

example 30 ms for the TCU), the allowed BS-TCU delay commonly shrinks to 10 ms. This corresponds in practice to an ATM route with no more than 30 ATM hops.

Beside the delay other technical constraints have to be considered, particularly resilience, alarm transmission or synchronisation issues. The migration of 2G traffic to ATM or IP-networks offers various economical opportunities as it contains various technical traps and risks depending on the complexity of equipment manufacturers, services and configurations. For this reason, the implementation strategy cannot be standardised but needs to be tailored exactly to the operator's circumstances.

17.2.2 MPLS-ARCHITECTURE FOR FUTURE 3G TRANSMISSIONS

The rapid growth of data volumes and services in the World Wide Web led to new QoS requirements for IP. Particularly the establishment of specific user groups or services with different requirements for availability, throughput and performance in one common IP-network are of a particular interest for mobile operators, since they apply commonly more complex tariff structures to their customers than IP-carriers. Most of the current IP technologies have limited potential for traffic engineering and aggregation. MPLS (Multi Protocol Layer Switching) addresses these requirements. Actually developed for IP networks, MPLS integrates the label-swapping paradigm with network layer routing. It further supports the delivery of services with QoS-guarantees, which can improve the price per performance of network-layer routing [26]. The main quality improvement of MPLS is depicted in Figure 17.13, which shows in particular the MPLS-ATM interworking (MPOA) as defined in [27].

MPLS encapsulates ATM or IP-Packets by adding a label in the Inter Working Function (IWF), which contains an LSP-Information (Label Switched Path) that is used by the LSR (Label Switched Router). ATM Packets of the same VC can be switched to different LSPs. Likewise, several VCs can be mapped to one LSP, which is a significant improvement for the efficient transport of broadcast information to multicast services. For IP-networks, MPLS is a significant step forward to multicasting as well as prioritised IP-routing, which minimises the end-to-end delay. If the IWF is implemented in the service unit (e.g. the Node B), then services with the same priority (like 2 voice services) can be routed or prioritised differently, which increases the options to establish QoS for instance to premium users.

One great strategic advantage to current 3G-operators, who deployed a significant amount of classic ATM switches, is the interoperability of MPLS over IP and MPLS over ATM, which allows the usage of the same routing protocols for IP, LAN or WAN sessions. The most common protocols are [9]:

- Transmission Control Protocol/Internet Protocol (TCP/IP);
- Routing Information Protocol (RIP);

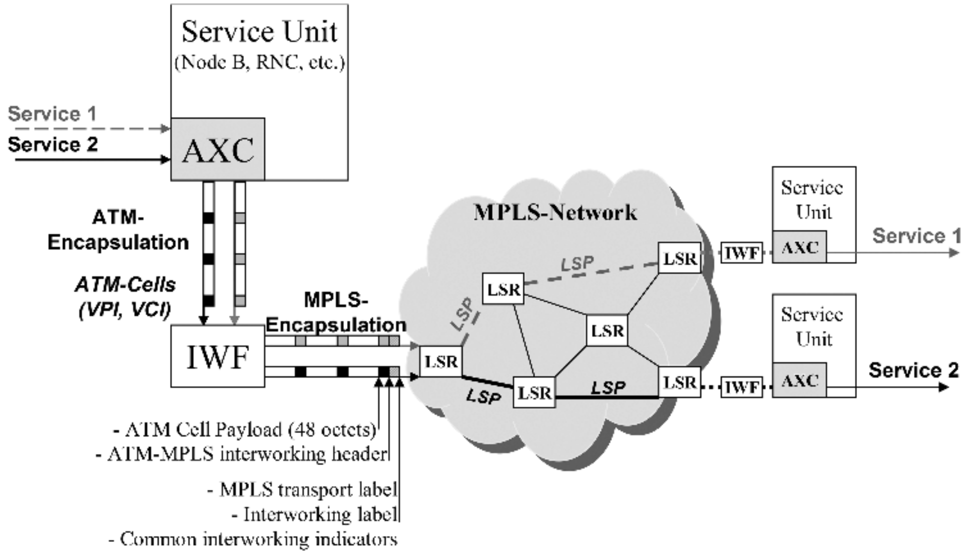


Figure 17.13 MPLS network structure.

- Open Shortest Path First (OSPF) or Multicast OSPF (MOSPF);
- Intermediate System to Intermediate System (IS-IS);
- Border Gateway Protocol-4 (BGP-4);
- Protocol Independent Multicast (PIM);
- Distance Vector Multicast Routing Protocol (DVMRP).

This also allows the implementation of end-to-end QoS-tools based on these protocols, particularly of Differentiated Services (DiffServ) or Integrated Services (IntServ). The interoperability of IP and ATM via MPLS could finally allow a smooth and successive migration from ATM/PDH/SDH towards IP on either PPP/SDL or LLC/SNAP on Ethernet. This will be described in the next section.

17.2.3 THE PATH TO DIRECT IP TRANSMISSION NETWORKING

A key learning of 2G deployment was the fact that building up a separate transmission infrastructure for GSM-voice and data (GPRS) was costly. Furthermore, it achieved a low performance given the invested infrastructure since backhaul capacities had to be kept separate on a shared physical link. For this reason, the main idea of 3G Release 04 was the convergence of voice and data networks, as shown in Figure 17.14 from IPv6-forum [28].

As described earlier, many operators implemented UTRAN R99 using classic ATM switches. MPLS networks can use traditional ATM equipment as a migration step in introducing MPLS to an existing ATM network. Traditional ATM switches can be used in three ways:

1. Backhauling, when the access device is remote from the edge LSR. The access device is connected to the edge LSR by PVCs switched through an ATM network.
2. Tunnelling through ATM switches between an edge LSR and an ATM LSR. In this case, the edge LSR does not need to be adjacent to an ATM LSR, but can be connected through an ATM network.
3. Tunnelling through ATM switches between ATM LSRs. In this case, the core network uses traditional ATM switches as well as ATM switches.

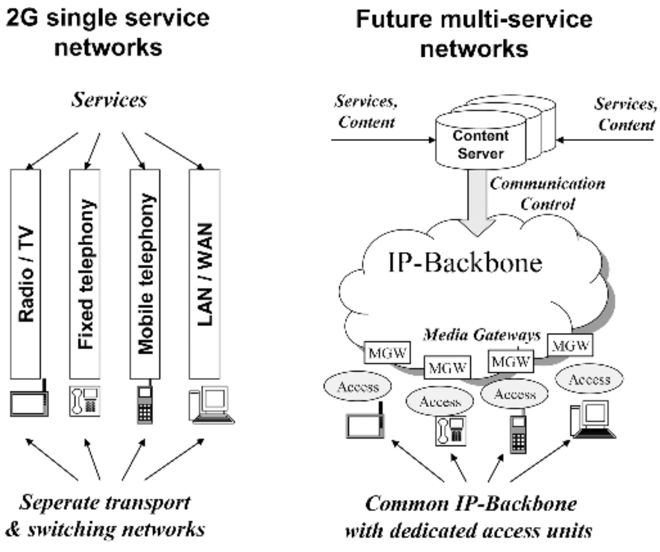


Figure 17.14 IP evolution scenario.

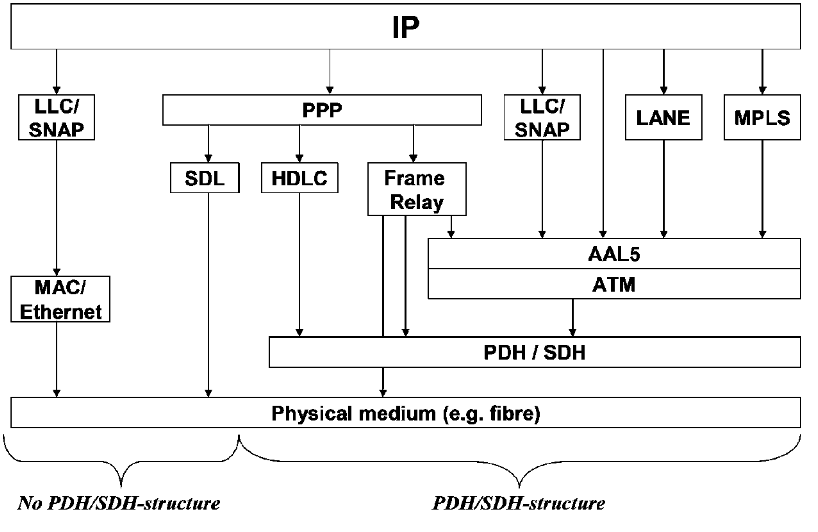


Figure 17.15 Transport protocols for IP-services.

The use of traditional ATM equipment for IP-services has the disadvantage of excessive overhead and higher delay [4,23,22]. The reason can be found in the protocol structure, which is depicted in Figure 17.15.

Each protocol layer adds overhead as well as end-to-end delay. As can be seen in Figure 17.15, the use of PPP/IP on FR on AAL5 on PDH incorporates more protocol layers and thus more overhead and delay compared to IP on Ethernet or SDL. Secondly, PDH based on ITU G703 is defined as symmetric for uplink and downlink. This is efficient for conversational services, but not at all for data streaming, online-TV or downloads.

On the other hand, many operators have implemented PDH or SDH-systems, and not MAC/Ethernet. For this reason, with the growth of IP-based services, a trigger for a migration to MAC Ethernet or SDL needs to be defined. Recent scenarios see such a trigger when at least 80 % of the transport traffic is IP-based or when the uplink/downlink asymmetry is more than 5:1. A feasible scenario is the successive migration of clusters from PDH to Ethernet by interoperating them via IP and MPLS.

A key driver for the time frame is still the development of voice and video telephony. Particularly, the recent success of VoIP services could decrease the time frame to a migration away from PDH/SDH towards MAC/Ethernet.

17.3 END-TO-END TRANSMISSION DIMENSIONING APPROACH

In classic 2G networks, the dimensioning of transmission networks has been a straightforward process based on the Erlang B formula. Since the backhaul systems were almost all circuit switched, the dimensioning had only to consider the air interface and the interface between the base station controller and the switch [23].

In 3G this changes considerably, due to the following reasons:

- In the Node B, not only the throughput can be limited by the air-interface due to cell breathing and interference, but also by the OSVF-codes (amount of Node B baseband units), which is a viable scenario when HSDPA is introduced.
- Secondly, the Node B ATM Cross Connect (AXC) allows overbooking of VCs in the ATM network.
- Thirdly, dynamic traffic routing like PNNI introduces new challenges on delay and transmission availability triggering new dimensioning requirements.
- Finally, new end-to-end services introduce additional overheads, retransmissions and delays that have to be considered on a dimensioning approach.

As long as no retrievals of utilisation and blockings of errors are available, an end-to-end simulation approach would start, section by section, with the Node B first, then with the ATM network, and finally with the services on top of it. Section 17.3.3 will therefore use IP-services as an example.

17.3.1 DIMENSIONING OF NODE B THROUGHPUT

As already described in Section 17.2.1.4, the most practical approach to dimension a Node B capacity requirement is the grouping of all services either into circuit switched and packet switched traffic or delay sensitive and non-delay sensitive traffic. To model the different services and data rates, an adequate multirate model has to be applied to the Iu-interface. To begin with the multirate model, we assume a mix of circuit switched services with rates at a granularity $n \cdot 16$ kbps; commonly, a voice service requires $1 \cdot 16$ kbps on a PDH/SDH frame. This is illustrated in Figure 17.16.

17.3.1.1 Dimensioning Methods

Generally, there are two methodologies to evaluate the required bandwidth [23]:

1. *Statistical approach* The services during a busy hour are modelled as calls with an individual data rate and data activity as shown in Figure 17.17. This approach can be executed with several resources (like Excel). On the other hand, it cannot model the time dependencies of protocols, and the effects of buffer sizes, latency and cell delay variation.
2. *Analytical approach* This approach models the protocols of the network elements involved. The effort of this method is quite high, since an in-depth knowledge of the vendor equipment is required.

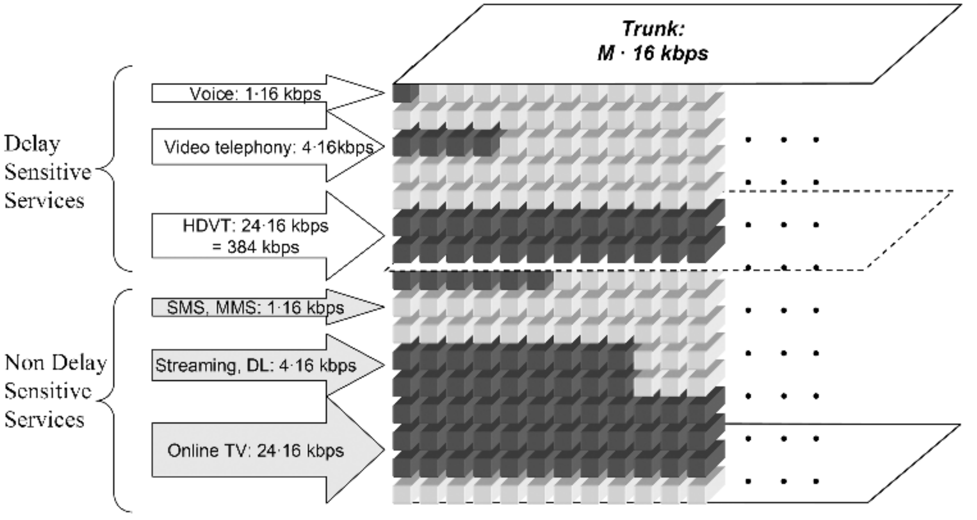


Figure 17.16 Dimensioning of Node B transmission resources.

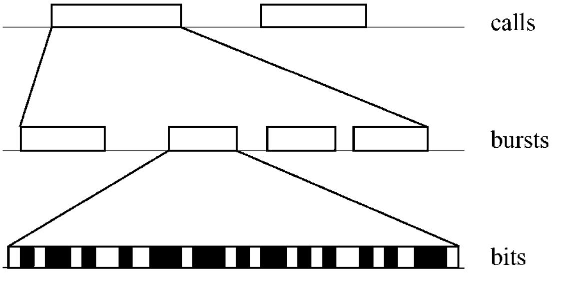


Figure 17.17 Statistic model for transmission dimensioning.

The suitable approach is then:

1. Prioritise delay sensitive traffic by calculating the required bandwidth on a busy hour first.
2. Calculate the required bandwidth of the non-delay sensitive traffic. Here, it should be distinguished whether the service requires a stringent delivery time or is a background service.
3. Calculate the required bandwidth for the Node B control channels. These are usually one channel for OAM, one for the Baseband Cards (D-NBAP) and one for the Node B radio modules (C-NBAP).
4. Add the ATM overhead. This will be done in Section 17.3.2.1.

The main factors for the traffic calculation are:

- User Data rate.
- Service activity factor: As a difference to the air interface the activity for circuit switched services has to be assumed as 100 %, except silence detection and suppression has been implemented in the Node B transmission unit.
- Burst efficiency: Most vendors support a sustained transmission of a FACH for 5–20 seconds on bursty services. The intention is to keep alive higher layer protocol sessions like telnet or TCP/IP.

This can be severe, if the service involves mainly the transmission of small packets like on web browsing or interactive gaming. In the worst case, 384 kbps FACH resources could keep allocated up to 20 seconds after the transmission of a 16k packet making the service extremely burst inefficient.
- Number of users using the service in a busy hour.
- Offered traffic (in Erlang) and required blocking (in %) for delay sensitive services.
- Retransmission percentage and maximum delay of 95 or 99 % packet quantile.
- Soft handoff overhead, if soft handoff is applied to the service.

All user channels as well as the Node B signalling channels (OAM, CNBAP, DNBAP) will then be ATM encapsulated. The required ATM overhead will be shown in Section 17.3.2.1.

The analytical approach for a Node B requires a higher effort. The example in the next section shows a FDD and TDD simulation of a number of users having the same user profile. As an example, the computational tool give the option to configure the number and length of sessions during the busy hour (BH), the data volume as well as the maximum FDD or TDD burst size. The principle is as follows:

1. Generate an offered traffic for each of the users for every second of the BH.
2. Simulate the absorption of the offered traffic given the configured FDD and TDD burst sizes for every second of the BH.
3. Optional: rerun the calculation with a varying number of users.

The example of the main configuring screen of the tool can be seen in the Figure 17.18.

In the tool, the limit for maximum throughput can be set to the limit at the RF side of the Node B. As will be seen, one of the most sensitive parameters when configuring the throughput for a service mix mainly consisting of data services, is the maximum burst size. This will be illustrated in the next section.

Offered Internet traffic

Number of users during a busy hour per cell	10
Internet user profile	
Number of sessions per user during a BH	2
Number of Web Pages per session	20
Data volume per Web site (kByte)	15
Number of Downloads per session	3
Size per Download (Mbyte)	6
Average online seconds per user during busy hour excluding data downloads	600
Number of Web Page changes per user per BH	40
Number of Downloads per user	6
Number of CS-sessions per user	2
Time per CS-second (s)	100
Data rate of CS-second (kbps)	16
Average download size per user during busy hour (MByte)	0.30

	TDD	FDD
Number of 5 MHz-carriers	2	1
Chiprate (Mbit/s)	7.68	3.84
Maximum burst rate per user (Kbit/s)	1536	768
Eb/No (dB)	5	5
Code Orthogonality	85%	85%
Frequency reuse efficiency	50%	50%
Downlink Pole capacity during a Downlink time slot (maximum number of users per cell using the maximum burst rate)	5.270462767	5.270462767
Load	60%	60%
Maximum throughput rate during a Downlink time slot	4857.258486	2428.629243
Number of Time slots per frame usable for downlink traffic (maximum 15)	9	15
Maximum Downlink throughput per second (Kbit)	2914.355092	2428.629243

Simulate ALL

Simulate Offered Traffic

Simulate TDD | Simulate FDD

Variate LOAD

From	10	user
Until	10	user
Intervalls	5	

Figure 17.18 Main screen of VBA dimensioning tool.

17.3.1.2 Example Calculations

Table 17.4 shows the result of the statistical approach calculations. Adding an additional Layer 2 overhead, which is commonly ATM & AAL1-5, the required bandwidth would increase at least another 30 %; this means that at least four E1 links would be required to transport the load in this scenario.

The results of the analytical approach show the burstiness of internet browsing and download compared to circuit switched sessions. Commonly, downloads contain file sizes of several Mbytes, which are offered (in theory) in a split of a second. The offered traffic graph in Figure 17.19 shows that the highest peaks appear whenever a download is requested.

It then depends on the maximum burst size of the transport unit how to absorb the sudden DL traffic load. From Figures 17.20 and 17.21, it can be seen that a maximum bearer burst size of 128 kbps requires a significantly longer time compared to a 384 kbps bearer.

The results show also that for Internet services a low maximum bearer size leads to a long download time, even when the total Node B resources are not congested. The higher bearer size can absorb bursty traffic much better, which leads to higher internet user satisfaction. Many classical telecom operators have configured bearer sizes of not more than 384 kbps to ensure congestion rates smaller then 5 %. This is a classical circuit switched approach, which dissatisfies Internet users. If an internet user could download a file on 512 kbps with an assumed 20 % of blocking, the total download time would still be faster compared to a download on 384 kbps and 1 or 0 % blocking. Surfing and download Internet users can tolerate blocking of 20 % during download time, which would, of course, be disastrous for circuit switched services.

This shows that Node B dimensioning heavily depends on the type of service transported, and that traffic cannot be regarded as traffic without the context to the applied service.

Table 17.4 Results of statistical Node B dimensioning.

User Spectrum Required

Delay sensitive-Service	1	2	3	4	5	6
Name	Voice	Video telephony	HDVT	Online Gaming		
Data rate (kbps)	12.2	64	384	12	0	0
Activity factor (%)	100%	100%	100%	5%	0	0
Burst efficiency	100%	100%	100%	10%	0	0
Number of users	35	12	2	200	0	0
Blocking (for DS-services)	0.20%	0.10%	0.10%	0.10%	0.10%	0.10%
Offered traffic (mErl) per user	23	15	10	20	0	0
SHO-overhead	35%	35%	35%	35%	0%	0%
Total traffic	0.81	0.18	0.02	2.00	0.00	0.00
Required Channels	5	4	2	8	0	0
Required Spectrum (kbps)	82.35	345.6	1036.8	129.6	0	0

Non Delay Sensitive-Service	1	2	3	4	5	6
Name	SMS, MMS	Internet surfing	Online TV	Download		
Data rate (kbps)	12	12	384	384	0	0
Activity factor (%)	10%	2%	80%	5%	0	0
Burst efficiency	50%	15%	100%	30%	0	0
Number of users	200	150	2	30	0	0
Retransmission percentage	10%	0%	0%	24%	0	0
SHO-overhead	0%	35%	35%	0%	0	0
Required Spectrum (kbps)	528.0	324.0	829.4	2380.8	0.0	0.0

User Spectrum required for Node B	5656.59	kbps

Spectrum for Signalling

Type of node B-signalling	CNBAP	DNBAP	OAM
Required Spectrum (kbps)	50.0	50.0	50.0

Complete Spectrum required for Node B (except ATM)	5806.59	kbps

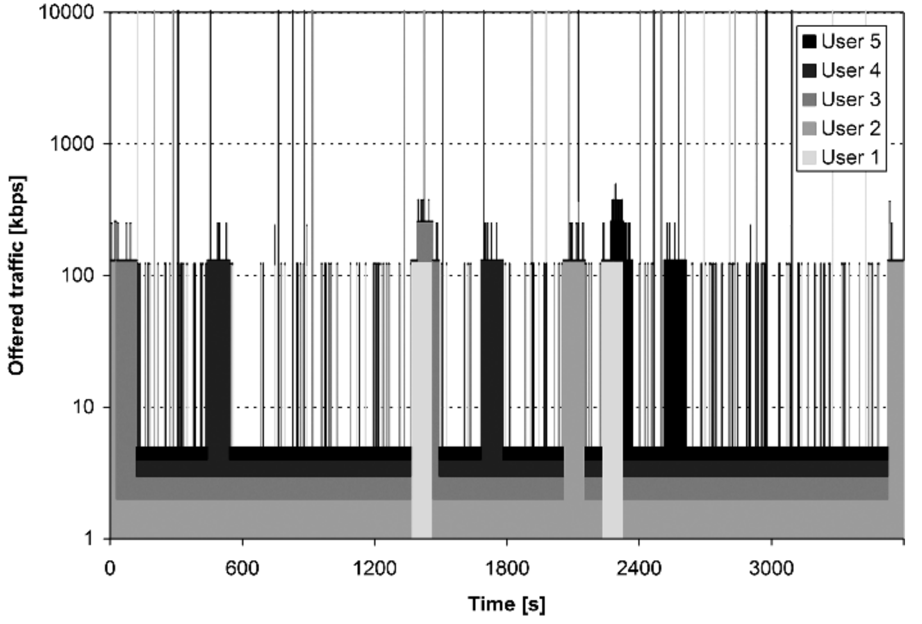


Figure 17.19 Offered traffic for data services during a busy hour.

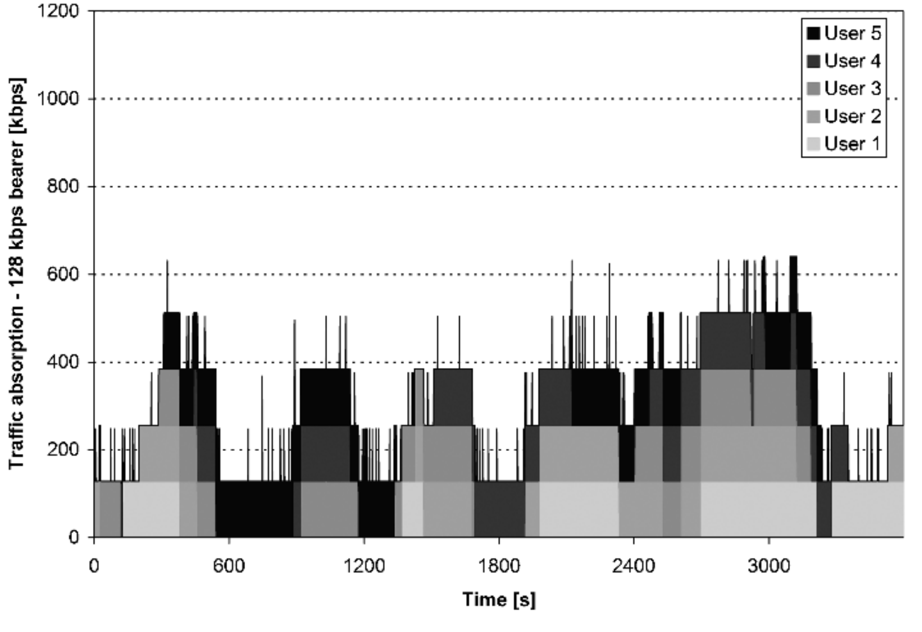


Figure 17.20 Offered traffic absorption for 128 kbps bearer.

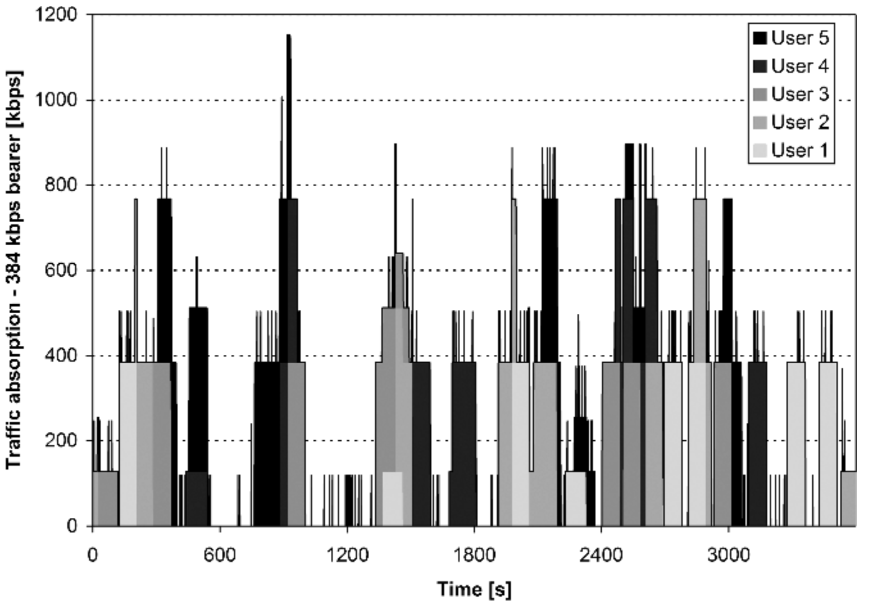


Figure 17.21 Offered traffic absorption for 384 kbps bearer.

17.3.2 TRAFFIC DIMENSIONING OF THE ATM NETWORK

As already mentioned, the Node B AXC allows overbooking of VCs in the ATM network. Secondly, dynamic traffic routing like PNNI introduces new challenges on delay and transmission availability. Section 17.3.2.1 illustrates the ATM overhead of all ATM adaptation layers.

While overbooking of an ATM network can be handled by linear multiplication of the Node B traffic, the handling of PNNI on a meshed network requires more complex formulas. Even more complex is the handling of buffer sizes and transmission delays on ATM networks.

17.3.2.1 Dimensioning Methods

The principle dimensioning methods for Node Bs, described in Section 17.3.1, apply also for the dimensioning of the ATM network. Table 17.5 shows the overhead to be considered for the ATM adaptation layers.

Except for AAL1m the maximum overhead can deviate significantly depending on the burstiness of the traffic. On AAL2, empty bytes of an ATM container are padded, which does not apply to AAL5. Additionally to this overhead, the AAL Signalling VC has to be added.

While the handling of ATM overhead can be considered straightforward, the consideration of PNNI in a ring or even a meshed network is rather complex. Section 17.3.2.2 shows an example of how to calculate this traffic.

Table 17.5 Minimum overhead of different ATM adaptation layers.

Adaptation type	AAL1	AAL2	AAL3/4	AAL5
Minimum overhead	16.40 %	17.78 %	20.47 %	10.50 %

Even more complex is an analytical approach, which considers a ring or meshed network topology as well as the ATM protocol structure and the configured buffer sizes of the network elements. Current simulation software is capable of simulating meshed networks, including 200 network elements with most common transport protocol structures. In comparison to air interface simulation, this shows that the analytical simulation of a transport network requires significantly higher resources.

17.3.2.2 Example Calculations

When applying the same example for the statistical approach of Section 17.3.1.2, ATM adds a significant overhead, particularly when the payload packages are small, e.g. during on-line gaming or web surfing. Compared to the initial payload, the requirement for the transport capacity would then increase from four E1 links to more than six E1 links. To ease complexity of the calculation of the network presented in Figure 17.22, equal link costing, transmission capacity and one peer group was assumed.

As already mentioned, the analytical approach is very complex, and requires enhanced IT capabilities. However, to give a hands-on approach for a fixed network planner, the mesh just illustrated has been used for an analytic link failure scenario, which is described below.

The example presented in Figure 17.23 shows the scenario of the described ATM cluster with a link failure on link number 4 after 1800 seconds during a busy hour. As can be seen, the traffic on link 4 has to be re-routed after 1800 seconds. As a result on the remaining links, the traffic increases by 25 % on average the moment the link number 4 fails. The exact amount depends on the link capacity, the costing and the existing traffic.

17.3.3 TRAFFIC DIMENSIONING OF THE IP-NETWORK

Like in ATM, it is important to know the type of service and its session protocols when dimensioning the overhead generated by IP. The most suitable adaptation layer for IP traffic in mobile networks is commonly AAL5 [29]. However, the most common adaptation layer in the mobile application world

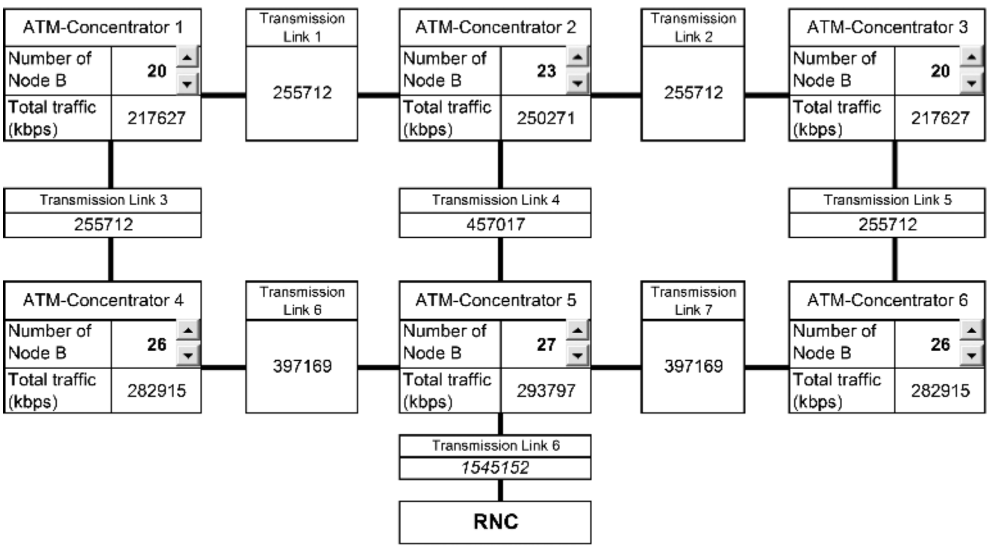


Figure 17.22 ATM dimensioning of a meshed network.

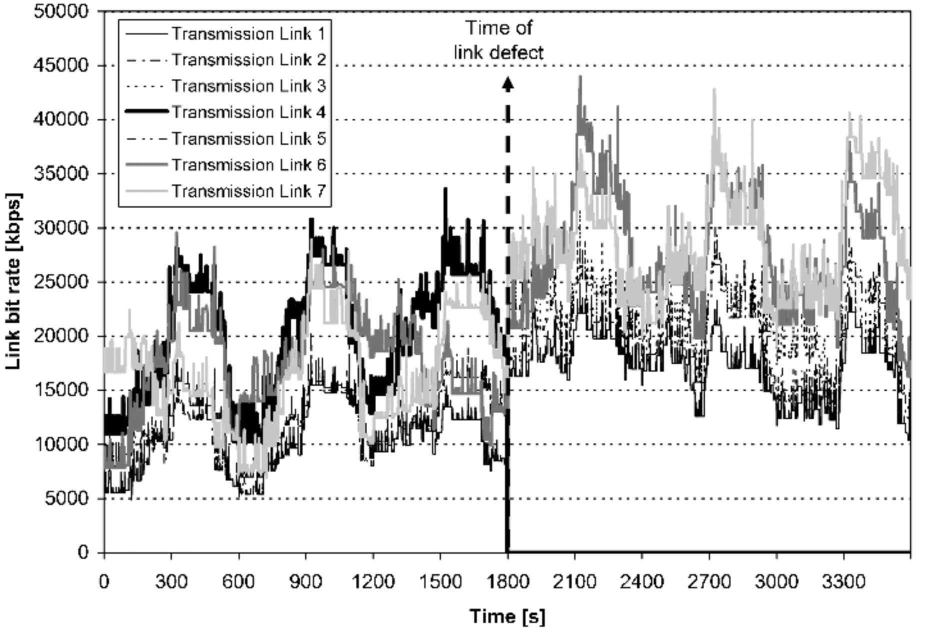


Figure 17.23 Bit rate in link 4 failure scenario in a meshed ATM network.

is AAL2, which increases the overhead. As will be seen in the next section, the main drivers for the overhead are the payload size in the IP-packet as well as the size of the Maximum Transfer Unit (MTU).

When dimensioning the IP overhead, the easiest approach is to work from the OSI application layers to the session layer, to the link layer, and finally to the physical layer. Table 17.6 describes the minimum overheads of the most common protocols used for the application and session layer [30].

Table 17.6 Overhead of IP protocols.

Type of protocol	Description	Minimum overhead
TCP	TCP adds a header of 20 bytes to the payload. TCP enables packet retransmission, multiplexing and data integrity	2.13 % on Ethernet G802, 0.36 % on Gigabit Ethernet
UDP, RTP	UDP or RTP adds a header of 8 bytes to the payload used for multiplexing and data integrity	0.53 % on Ethernet G802, 0.09 % on Gigabit Ethernet
IPv4	IPv4 encapsulates the payload into a 20 byte header. The IPv4 address has only 4 byte. The IPv4 allows refragmentation in a transmission router.	1.33 % on Ethernet G802, 0.22 % on Gigabit Ethernet
IPv6	Ipv6 encapsulates the payload into a 40 byte header. The IPv4 address has 8 byte. The IPv6 does not allow refragmentation in a transmission router	2.61 % on Ethernet G802, 0.44 % on Gigabit Ethernet

In contrast to IPv4, IPv6 does not provide for fragmentation and reassembly. If an IPv6 packet received by a router is too large to be forwarded over the outgoing link, the router simply drops the packet and sends a 'Packet Too Big' ICMP error message back to the sender. The intention is to optimise the end-to-end delay of the IP network. The sender can resend the data, using a smaller IP packet size. Fragmentation and reassembly is a time-consuming operation; removing this functionality from the routers and placing it squarely in the end systems considerably speeds up IP forwarding within the network. If the packet has been transmitted by UDP, retransmission is not executed. This configuration ensures an acceptable small overhead, and is optimal for streaming, but not for downloads.

The protocol encapsulation on the link layer leads to the minimum overhead, as displayed in Table 17.7. Finally the overhead imposed by the physical transmission medium has to be considered, as shown in Table 17.8.

Tables 17.9 and 17.10 present an example tool with input information as provided in the previous Tables 17.6 and 17.8.

Figure 17.24 illustrates the main driver for the overhead to the IP payload. It can be clearly seen that the overhead generated by IP depends not only on the protocols used, but particularly on the

Table 17.7 Overhead of Layer-2 protocols.

Type of protocol	Description	Minimum overhead
LLC/SNAP	LLC adds an additional 8 byte header to each packet. The packet does not have to exceed the MTU-size	0.09–17.7 %, depending on MTU
ATM	The protocol structure has been explained in Section 17.2.1.1. If AAL5 is applied the ATM cell will eventually be padded, if there are still free octets after the 8 byte AAL5 trailer	11–40 %, depending on AAL type and payload size

Table 17.8 Overhead of fibre transmission systems.

Type of protocol	Description	Minimum overhead
Ethernet G802	Ethernet adds 38 byte of overhead plus 4 byte, if the optional 4 byte VAN Tag is used. The maximum Ethernet payload is 1500 bytes	Ca. 2.27 %
Gigabit Ethernet	Gigabit Ethernet has the same overhead structure like Ethernet G802, but a payload size of 9000 bytes	Ca 0.46 %
Sonet OC-3 or OC-12	Sonet OC-3 has a 2430 byte frameout of which 90 bytes are overhead. Sonet OC-12 has a 9720 byte frame with a 360 bytes overhead	Ca. 3.77 %
Multimode Fibre 100 Mbps	A multi mode fibre adds a 2 byte overhead between a 53 byte ATM cell	Ca. 3.95 %
DS3	A DS3 has a 44736 byte frame, out of which 40704 bytes are payload	Ca. 9.97 %

Table 17.9 Example of dimensioning sheet for IP Ethernet.

	Gigabit Ethernet with Jumbo Framing	Ethernet G.802
User Payload-Size per Data-packet	9000	1500
Application Protocol	TCP	TCP
Overhead due to application protocol	0.36%	2.13%
IP-Version	v6	v6
Overhead due to IP	0.44%	2.61%
MTU-Size	6134	6709
LLC/SNAP-Overhead	0.18%	0.12%
Ethernet G802-Tagging	☑ VLAN-Tag (Y/N)	☑ VLAN-Tag (Y/N)
Overhead due to Ethernet	0.46%	2.66%
Total overhead	1.44%	8.13%
Physical line Rate (Mbps)	1000	100
Maximum User line Rate (Mbps)	985.76	92.48
Transmission Rate after 4B/5B-encodding acc. To FDDI	1250	125

Table 17.10 Dimensioning sheet for IP on SDH and PDH.

	STM-1 or OC-3	STM-4 or OC-12	MM-Fibre 100 Mbps	DS3
User Payload-Size per Data-packet	1000	600	400	200
Application Protocol	TCP	TCP	TCP	TCP
Overhead due to application protocol	3.20%	5.33%	8.00%	16.00%
IP-Version	v4	v4	v6	v6
Overhead due to IP	1.98%	3.29%	9.26%	17.24%
MTU-Size	1500	1500	1500	1500
LLC/SNAP-Overhead	0.78%	1.27%	1.69%	2.94%
AAL-Mode	5	5	2	2
Overhead due to ATM	12.55%	16.67%	21.46%	32.50%
Overhead due to physical medium	3.77%	3.77%	3.95%	9.97%
Total overhead	21.00%	28.33%	51.50%	104.00%
Physical line Rate (Mbps)	155.52	622.08	100.00	44.74
Maximum User line Rate (Mbps)	128.53	484.74	66.01	21.93

payload of the datagram of the IP-packet payload. For payload sizes less than 50 bytes, the overhead introduced by the protocols increases to more than 250 %. This is severe for interactive gaming, ping, SMS or any small band transport traffic.

The notches in the graph for all ATM protocols are due to the padding imposed by the ATM adaptation layers. For packet sizes bigger then 200 kB, the limits of ATM-based IP compared to IP on Ethernet are very transparent. The highest overhead on high packet sizes is on DS3 making it the medium of last choice. The graph is based on an MTU size of 576 bytes, which is the Internet network default. For higher MTU sizes, Gigabit Ethernet can utilise its higher packet size.

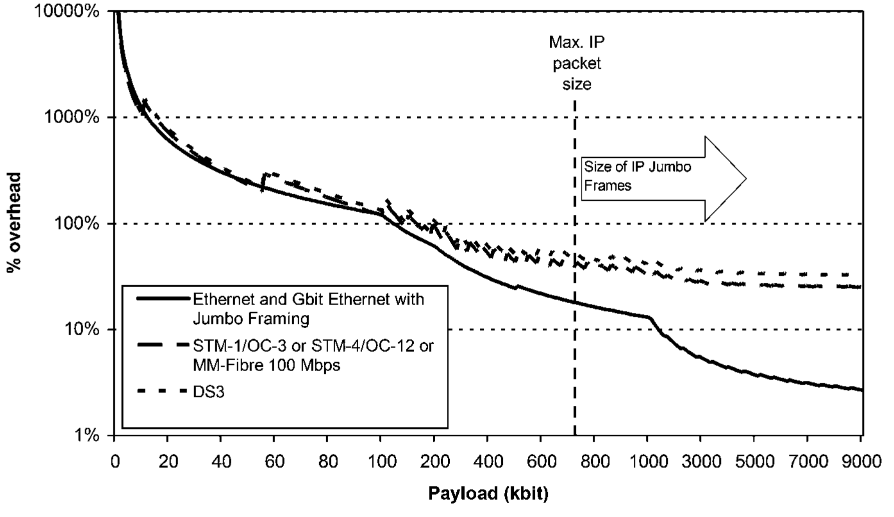


Figure 17.24 Overhead dependency to IP packet payload size.

17.4 NETWORK SOLUTIONS FOR UTRAN TRANSMISSION INFRASTRUCTURE

17.4.1 LEASED LINES

Leased lines (LL) are a main type of circuit-switched WANs, which allow permanent connection between two points set up by a telecommunications common carrier, therefore sometimes referred to as private lines or dedicated lines. Businesses, service provider network operator and even individuals typically use leased lines to connect geographically distant facilities (i.e. offices, network devices) or for Internet access. There are two types of the leased lines [31]:

1. Point-to-point leased lines;
2. Multi-point or multi-drop leased lines.

The point-to-point leased line is used to connect two separate facilities directly one to another, providing full-time and full-capacity communications. Leased line point-to-point channels are often terminated into multiplexers that connect to other telecommunication devices.

The multi-point or multi-drop leased lines are used to join several different facilities to a single central facility with common transmission channels. The devices attached to those channels share the capacity of the channel. Using a polling protocol, the central facility controls each attached device for occurrence of data traffic. When the pooled device has data to send, it transmits the data. Otherwise, it sends a short response signal to the central facility to enable the polling of the next device onto the multi-point channel. The central facility receives data inputs from all remote devices, prepares responses to the data inputs, and transmits the responses to the appropriate multi-point device. Both types of leased lines can be accomplished by using a number of approaches with throughputs from 64 kbps to 622 Mbps and more:

- Digital Data Services (DDS), which essentially use 56 or 64 kbps channels;
- ADSL which is currently offered using 768, 1536 or 3072 kbps;
- T-1 (DS-1) with bit rate 1.544 Mbps or T-3 (DS-3) with 44.736 Mbps carrier backbone;

- E-1 (CEPT-1) with bit rate 2.048 Mbps, E-3 (CEPT-3) with bit rate 34.368 Mbps or E-4 (CEPT-4) with bit rate 139.246 Mbps carrier backbone;
- J-1 with bit rate 1.544 Mbps or J-3 with bit rate 32.064 Mbps carrier backbone;
- SDH/SONET (Synchronous Optical Network Technologies) backbone:
  - STM-0/STS-1 (OC-1) with bit rate 51.48 Mbps
  - STM-1/STS-3 (OC-3) with bit rate 155.52 Mbps
  - STM-3/STS-9 (OC-9) with bit rate 466.56 Mbps
  - STS-12/STM-4 (OC-12) with bit rate 622.08 Mbps;
- *Dark fibre* backbone (customer leases the fibre itself), using commonly Ethernet like G802 or Gigabit Ethernet.

To set up the leased line, the transmission media and Channel Service Unit (CSU) are needed. The transmission media might be copper twisted-pairs or fibre with higher capacities. The CSU terminates each end of the carrier facility (e.g. E-1, T-1). The CSU equalises the received signal, filters the transmitted and received waveforms, and interacts with another customer's devices to perform diagnostics. Essentially, the CSU is used to set up the T-1/E-1/J-1 line with a customer-owned PBX, channel banks as stand-alone devices, intelligent multiplexers (for example, T-, E-, or J-carrier multiplexers), and any other DS-*x*/CEPT-*x*–compliant DTE, such as digital cross-connects. Using intelligent multiplexers, it is possible to manage transmission resources and to aggregate onto a higher-speed transmission line.

Leased lines provide high throughput and can come with SLAs (Service Level Agreement) and SLGs (Service Level Guarantee) that govern the 'uptime' of the circuit and technical support of the telecommunications provider.

On the benefit side, leased lines are required to support large networks, high speed private circuits, host web servers, transfer large amounts of data and files and run multimedia applications. Especially new 2G and 3G mobile operators need leased lines to complete their own backhaul networks.

The main disadvantages of this solution are the highest cumulative costs (CAPEX plus OPEX) as well as limited implementation. The CAPEX depends on installation and activation expenses. The OPEX is mainly connected with leasing expenses, which depend on the country, the operator, the geographical location, the transmission rates and the length of the leased line. Annual rentals for 2 Mbps at the distance of 2 km national leased line amounts to about a few thousands Euro a year. For instance, in Poland, the rentals for LL from the national telecomm operator are regulated by the decision of the president of the national telecommunications regulator (UKE). In other European markets, like UK or Germany, several city carriers have started to offer 3G backhauling using Ethernet G802, which is marketed as 'Metro Ethernet'.

Furthermore, the technical means of an LL customer to ensure SLA-guarantees are very limited, because most of the LL carriers disclose neither the route or used medium, nor the methods of how unavailable seconds are retrieved. This problem will increase with the rapid growth of Ethernet-products, which tolerate significantly lower availability standards for circuit emulated leased lines (see Section 17.2.1.6).

The limited implementation of leased lines to UTRAN transmission infrastructure concerns limited access to leased lines at many base station locations, especially out of urban areas. In many cases it requires additional investments that increase the CAPEX. Additionally, mobile operators using leased lines become dependent on other telecommunication operators; however, this is not always avoidable.

17.4.2 POINT-TO-POINT SYSTEMS

One of the main and the most popular elements of all backhaul and backbone networks are point-to-point (PTP) systems. They provide high capacity and high availability connections between two separate fixed facilities. For mobile cellular networks, the most flexible and cost-effective solutions are fixed-wireless broadband systems called radio relay systems or radio links. They operate with microwave frequencies (from 2 GHz up to 60 GHz) and match the capabilities of cable-based transmission systems as well as provide the same type of protocols and interfaces as leased lines (from E1 to STM-1). Radio relay systems enable outstanding generic benefits in planning and installing new backhaul networks. Fixed wireless relay systems are suitable solutions over difficult terrain, in rural areas or in old city centres where costs of wireline infrastructure or building restrictions may prevent from new construction [32]. For these reasons, the radio relay systems are being used as links between RNCs and Node Bs, between RNC and MSC or between RNCs. Their range and capacity depends on the used frequency band, the transmitter power, the transmit and receive antenna gains, the receiver sensitivity and the radio channel bandwidth with capacity ranging from a single E1 to two times STM-1 (311 Mbit/s).

Typical radio relay systems consist of two terminal stations (end points), which are very close to the source or destination of traffic. The terminal station is equipped with a set of identical radio transmitters, a set of identical radio receivers, filters, a directional antenna and multiplexer. Transmitters and receivers are connected to the shared antenna by coaxial cable or waveguide. The terminal stations can be divided into two parts: an indoor unit (IDU) and an outdoor unit (ODU). The indoor unit is connected with the traffic source or the traffic destination and supports some ODUs. The outdoor unit is integrated with antenna and mounted on a mast or a high tower. Parabolic reflectors, shell antennas and horn radiators are typical antenna types used in radio relay systems. Their diameter can reach up to 4 m. High antenna directivity permits efficient use of the available spectrum and long transmission distances, even when the transmit power equals to only 1 W.

Backhaul networks involve a very high availability and reliability, which typically ranges from 99.9 to 99.999 % of the time. The radio relay system availability is strongly connected with propagation conditions, deep fading and device malfunctions. To obtain the best propagation conditions, the terminal stations are located on hill-tops, mountain-tops, very high buildings and towers to achieve line of site (LOS) conditions with the first Fresnel zone free of obstacles (for more details, see Chapter 5). For the LOS propagation in the lower atmosphere strata (troposphere), the radio wave propagation is only affected by changes in barometric pressure, temperature, snowfall, rain, water vapour, turbulence and stratification (but of course the problem with multipath propagation still exists). For the frequencies above 10 GHz, scattering on hydrometeors and the molecular absorption are of great importance. For these reasons, the service range of radio relay systems depends on the climate zone where devices are to be installed. Figure 17.25 shows an example of the range of a radio relay system installed in regions with different rain intensity.

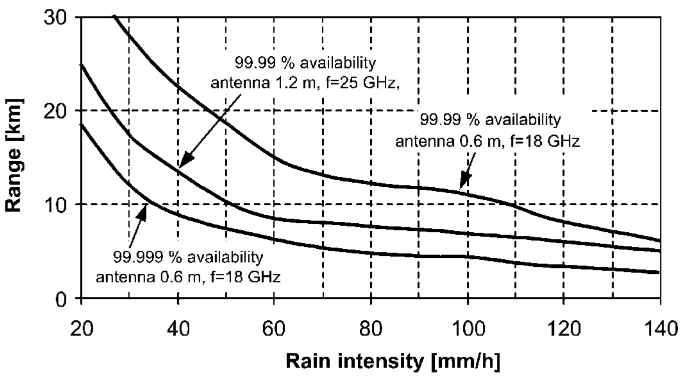


Figure 17.25 Example of the radio relay link ranges.

In the frequency range below 10 GHz, multipath propagation is considered as a dominant factor, which limits the range of the radio relay systems. The obstacles in the first Fresnel zone as well as atmospheric effects lead to problems of signal interferences and distortions (amplitude, frequency and phase dispersion). Only adaptive channel equalisation and diversity techniques are the best way to reduce multipath propagation effect. The multipath fading is overcome by using frequency diversity or space diversity. In frequency diversity the same information is transmitted by two separate links, which operate simultaneously on different frequencies (radio channels). In space diversity, additional antennas and receivers are required. These antennas have to be mounted at the same terminal point, but separated by a distance equivalent to some wavelengths.

Usually, the radio relay system capacity is not concentrated in a single channel but divided into some channels. Each channel uses an independent set of transmitter and receiver. In configurations without redundancy, all the sets are being used for data transmission between terminal points, providing maximum link capacity. In configurations with redundancy, only a part of the sets is used for transmission. The remaining sets are capable to take over any of the other sets in case of its malfunction. A typical configuration for radio relay systems is know as an '$m + 1$', where there is only one reserve radio set and m active sets [33].

Obstacles, the curvature of the Earth, the area configuration and reception issues have to be taken into consideration before radio relay links can be established. A single link can be established if the terminal stations are close enough so that the Earth's curvature can be neglected and if there is no obstruction on the radio path. Long distance relay links may require intermediate stations (repeater or relay station) to receive and retransmit the signals. Links between terminal station and relay station or between relay stations are called hops.

Large-scale mobile networks demand branching stations to concentrate the traffic from base station to the controller. Branching stations allow to comprise individual low capacity links into the one higher and multiple capacity link and arrange star-topology networks with limited antenna numbers in the RNC. An example of a UMTS radio infrastructure with radio relay links is shown in Figure 17.26.

Microwave radio relay systems are used to connect backhaul traffic at the local level or at the access level with capacity ranging from a single $n \times$ E1 to double STM-1 (311 Mbit/s). Utilised frequencies allow a connection over a link longer than 30 km. The access distances are typically short and capacity ranges from a $2 \times$ E1 up to $16 \times$ E1. At the local level, distances might be longer and the capacity requirement is larger, ranging from 16 E1 to STM-1. The fixed-wireless broadband systems deployed in 3G systems are a mix of SDH (Synchronous Digital Hierarchy) and PDH (Plesiochronous Digital Hierarchy).

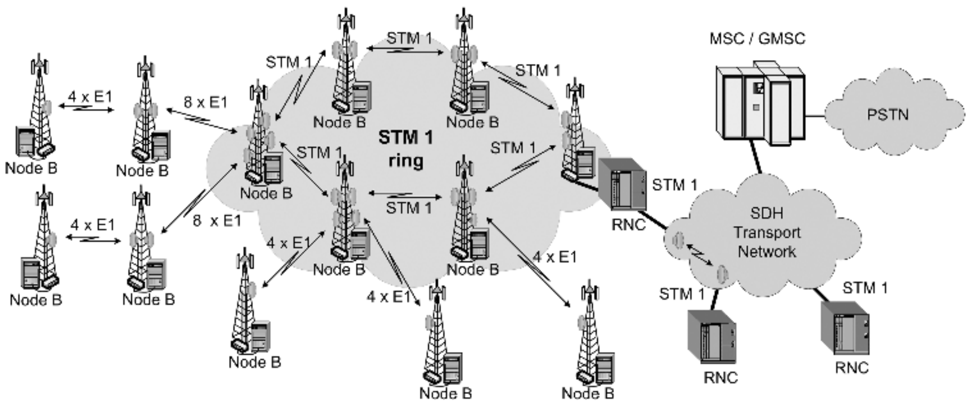


Figure 17.26 An example of UMTS backhaul network.

17.4.3 POINT-TO-MULTIPOINT SYSTEMS – LMDS

In metropolitan areas, high capacity cellular networks require a large number of base stations. All the base stations are usually located within a few hundred meters from each other. In order to serve the high densities of UMTS base stations, a large number of links to the RNC has to be established. In this case, the range of point-to-point microwave radio lines is not a serious problem; however, the number of links and required antennas on rooftops as well as costs and lose of flexibility constitute a problem. The Point-to-MultiPoint (PMP) system seems to be a better solution for the backhaul network to reduce costs and simplify network installations and future extensions. A single PMP base station can serve many remote Node Bs. Within the range of PMP entities, an operator has only to equip Node Bs with appropriate terminal stations and configure transmission parameters at the base station [32].

The radio transmission in PMP systems is not symmetrical as in PTP, and also line of sight (LOS) conditions are required. In the downlink (from PMP base station to remote terminals), the PMP base station broadcasts signals to all served entities in the cell or sector. The capacity of the broadcast channel is split and dynamically allocated between all the remote terminals. Each terminal is able to receive only information from broadcasted signals, which is dedicated to it. In the uplink, radio resources have to be shared between all the active terminals.

Due to the asymmetric transmission in uplink and downlink, the PMP systems are called Local Multipoint Distribution System (LMDS). The acronym LMDS is derived from the following:

- **L** (local) denotes that propagation characteristics limit the potential coverage area of a single cell site; ongoing field trials conducted in metropolitan centres place the range of an LMDS transmitter at up to a few kilometres;
- **M** (multipoint) indicates that signals are transmitted in a point-to-multipoint or broadcast method; the wireless return path, from subscriber to the base station, is a point-to-point transmission;
- **D** (distribution) refers to the distribution of signals, which may consist of simultaneous voice, data, Internet and video traffic;
- **S** (service) implies the subscriber nature of the relationship between the operator and the customer; the services offered through an LMDS network are entirely dependent on the operator's choice of business.

At the beginning, LMDS systems provided simple services, e.g. POTS data transmission, with low throughput as an alternative for point-to-point systems. At present, dependent on the configuration, LMDS systems provide a full range of broadband services with ATM or IP transmission for the last mile. The LMDS system architectures, access techniques, modulation methods as well as solutions are discussed in following sections.

17.4.3.1 System Architecture

LMDS systems are being designed for transport of multimedia services. This requires detailed technical specification for the system architecture to provide simultaneous access of many users to different services with defined quality of service and system availability. Special demands apply to an access and transmission network to allow dynamic transmission resource assignment on demand. A standard LMDS setup has a central facility connected to the operator's network hub via point-to-point microwave links. Basically, the point-to-multipoint wireless system architecture consists of four parts:

1. Network operations centre (NOC);
2. Fibre-based infrastructure;
3. Base station;
4. Customer Premise Equipment.

The network operations centre includes the network management equipment for managing regions of customer networks and can be interconnected with other NOCs. The fibre-based infrastructure basically consists of SDH or SONET links, the ATM and IP switching systems, interconnections with the network and the central office equipment.

The conversion from fibered infrastructure to wireless infrastructure happens at the base stations. The base station provides interface to fibre termination, modulation and demodulation functions, microwave transmission and reception equipment, as well as optionally local switching, which allows communications for customers without entering the fibre infrastructure. This function implies that billing, channel access management, registration and authentication have to be implemented locally within the base station.

The customer premise equipment varies widely from vendor to vendor. All configurations include indoor digital equipment and outdoor-mounted microwave equipment.

The system architecture depends on the type of delivery media to transport rendered services. The typical broadcast LMDS network architecture for Digital Video Broadcasting (DVB) networks with an interactive channel is presented in Figure 17.27. The interactive system is composed of Forward Interaction path (downstream) and Return Interaction path (upstream) [34]. The customer premise equipment, which is called Set Top Box (STB), consists of Network Interface Unit (NIU) and Set Top Unit (STU). The NIU comprises two network-dependent elements: Broadcast Interface Module (BIM) and Interactive Interface Module (IIM). The general concept is to use downstream transmission from the Interactive Network Adapter (INA) to the NIUs to provide synchronisation and information to all NIUs. This allows the NIUs to adapt to the network and send synchronised information upstream. The STB is equipped with interfaces to the broadcast and interactive channels, but only module IIM enables STB communication with the network.

Upstream transmission is divided into time slots, which can be used by different users. One downstream channel is used to synchronise upstream channels. A counter at the INA is sent periodically to the NIUs, so that all NIUs work with the same clock. This gives the opportunity to the INA to assign network resources to different users.

Three major access modes are provided with this system. The first one is based on contention access, which lets users send information at any time with the risk to have a collision with other users' transmissions. The second and third modes are contention-less based, where the INA either provides a finite amount of radio resources to a specific NIU, or a given bit rate requested by a NIU until the INA stops the connection. These access modes are dynamically shared with collision avoidance for the contention-less based access modes.

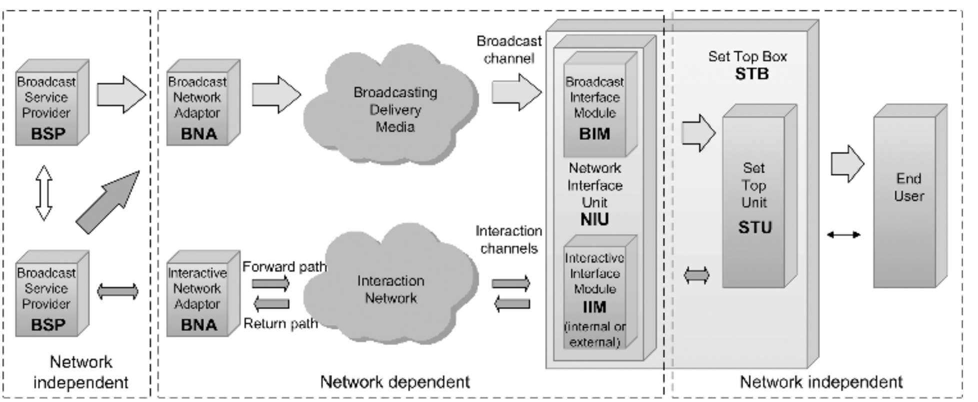


Figure 17.27 A generic LMDS reference model for interactive systems (based on [34]).

By periodically leaving a large time interval, the INA indicates to new terminals the possibility to execute the sign-on procedure, in order to synchronise their clock to the network clock without collisions with already active STBs.

All STBs have to support at least one of those solutions. Each of them differs in the overall system architecture, but provides the same quality of service. Both can be implemented in the same system under the condition that different frequencies are assigned to each system.

Nowadays, the complete implementation of the generic model presented in Figure 17.27 in interactive terrestrial broadband radio systems is too expensive and there is not any reason to do that in one system. Terrestrial broadband radio access systems can be split into two categories. The first category fulfils demands of telecommunication operators and the second one meets the requirements of cable television operators, who consider provision of telecommunication services. However, the interactive terrestrial broadband radio systems have immense competitors in various satellite systems. For this reason, as well as operators' demands and some technical aspects, mainly the terrestrial data transmission networks are being developed and implemented. The architecture of these systems corresponds with point-to-multipoint systems, the configuration of which depends on the used radio access methods. The radio is often the ideal way of obtaining communications at low cost, large distances and almost independent of the difficult topography.

According to ETSI EN 301 213-1 [35], a PMP system is comprised of a base station, which is called Central Station (CS), and a number of terminals (Figure 17.28). The CS may be subdivided into two Central Controller Station (CCS) and Central Radio Station (CRS) or more units. The CCS enables connection to the telecommunication networks via SNI interface and controls at least one or more number of CRS. The CRS provides the air interface to the terminal station or to repeater stations. Each CRS's radio transceiver is connected to a separate directional sector antenna to increase the capacity of the PMP system and to connect to other repeater stations or terminal stations within the CRS in MultiPoint–MultiPoint (MP–MP) systems.

The performance of the LMDS system transmission can only be achieved for an unobstructed line of sight between radio stations or terminals, e.g. CRS and Terminal Stations (TS). Repeater Stations (RS) are deployed to expand the range of the PMP network or to establish radio communication to some TSs without LOS in the direct radio path. The number of served TSs and RSs depends on the assigned channel bandwidth and provisioned services. The terminal station provides the interface to the terminal equipment, the subscriber premise equipment. A RS can serve one or more TSs or other

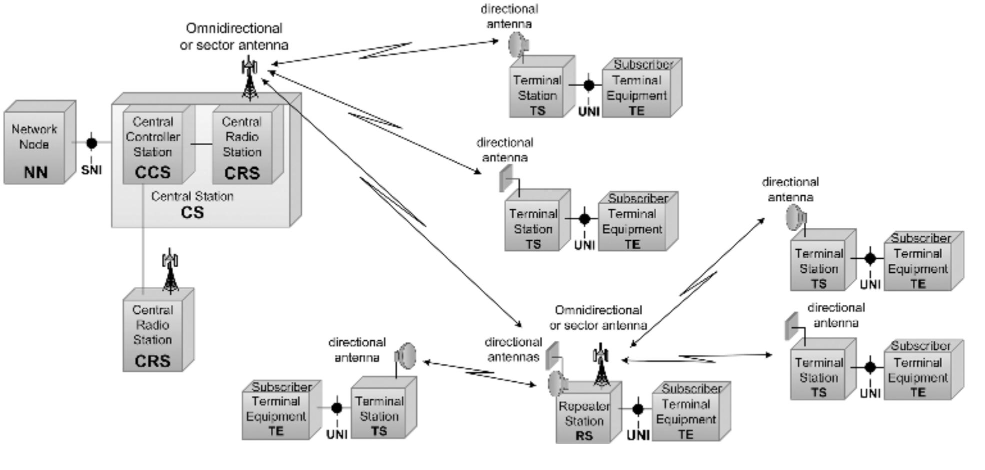


Figure 17.28 General PMP system architecture (based on [35]).

RSs. The RS may also provide the fixed interfaces to the Terminal subscriber Equipments (TE). The UNI is the point of connection to the subscriber equipment TS. All system elements are not necessarily deployed in a particular network, but the single CS is indispensable. In PMP systems, radio terminals from different manufacturers are not intended to interwork at radio frequency. Network operators have to choose solutions according to performance and availability requirements in order to extend the possible area of application, thus fitting to their network needs.

17.4.3.2 Access Techniques and Modulation Methods

PMP wireless broadband access systems use different access methods, taking into account the basic physical parameters of frequency, code and time. This leads to the four access methods of:

1. Frequency Division Multiple Access (FDMA);
2. Time Division Multiple Access (TDMA);
3. Code Division Multiple Access (CDMA);
4. Multi-Carrier Time Division Multiple Access (MC-TDMA).

The FDMA PMP system transmits a RF-signal from the Terminal Stations or Repeater Stations to the Central Radio Station only utilising a spectral bandwidth corresponding to that capacity which is requested from and assigned to the customer by Pre-Assigned Multiple Access (PAMA) or by Demand Assigned Multiple Access (DAMA). The Central Radio Station receives from each TS a single modulated carrier being processed independently within the CRS [36]. Thus, the CRS is receiving an FDMA PMP signal. For the FDMA links, the terminal station is allocated a fixed bandwidth, or a bandwidth varying slowly over time. FDMA access links fit in well if the user requirement is constant bandwidth and continuous availability.

In TDMA PMP systems, a central station broadcasts information to terminal stations in a continuous Time Division Multiplex (TDM) or in a burst TDMA mode [37]. The Terminal stations transmit in TDMA mode. The users may have access to the spectrum by sharing it through time multiplexing. The TDMA mode makes sense for customers who do not have a very heavy upstream traffic and just needs a 10 BaseT port for access to the Internet.

The DS-CDMA Central Radio Station transmits simultaneously and continuously to all active Terminal Stations within its coverage area, utilising a specific set of codes allocated to each active Terminal Station [38]. The terminal stations use the same, or a different, set of codes when transmitting to the CRS. Transmissions from CRS to TS are distinguished from transmissions on the other directions by using different frequency channels (FDD mode) or different time slots (TDD mode). Repeater stations may be placed for cell coverage enhancing. But CDMA supports a significantly smaller number of users than TDMA.

It is also possible to use Multi-Carrier Time Division Multiple Access (MC-TDMA) as a more flexible alternative to TDMA, FDMA and CDMA. The CRS and/or TS may transmit one or more sub-carriers at various frequencies, bandwidths, modulation and power levels. In these multiplexing access modes, the normative requirements relating to channel bandwidths and spectrum masks must be met. The terminal stations transmit in TDMA mode. The users may have access to the spectrum by sharing it through time multiplexing.

All of the abovementioned multiple access methods are implemented in the uplink channels (from TS or RS to CRS). In the downlink channels, most companies supply Time Division Multiplexing (TDM) or a burst TDMA mode.

The data rate capacity and the bandwidth spectrum efficiency of LMDS systems depend on the used modulation schemes. Modulation methods for broadband wireless LMDS systems are generally separated into Phase Shift Keying (PSK) and amplitude modulation (AM). Broadband transmission is only possible using higher order modulation schemes; for example, in LMDS systems with FDMA, BPSK, QPSK, DQPSK, 8PSK, 4QAM, 16QAM or 64QAM modulation methods can be used. Typical modulation schemes for LMDS systems in the downlink channels are QPSK, 16QAM or 64QAM. In the uplink channels, most applied modulations are QPSK and 16QAM. The 64QAM scheme is identified as optional due to the cost impact of supporting this format, resulting from stringent constraints on power amplifier linearity and radio phase noise requirements.

The Reed-Solomon forward error coding (FEC) and convolution coding methods protect radio transmission. Coding methods are combined with modulation schemes and are changed simultaneously to maximise the capacity of the available bandwidth for each radio channel condition. In particular, terminal stations located near to the base station could take advantage of 64QAM even during a rainstorm, while subscribers at the edge of a cell would be limited to QPSK.

The selection and implementation of an appropriate modulation scheme is driven from two primary perspectives, the first being the desired efficiency of the system and the second being the reliability of the network.

17.4.3.3 Capacity

The capacity of LMDS networks can be measured in terms of data transmission rate and maximum number of associated terminal stations (customer premises). For data rate calculations, the capacity of each LMDS base station has to be known. The LMDS base station capacity depends on its configuration, e.g. the number of sectors, types of used modulation schemes, sectors' overlapping areas, available channel number in the sector and channel bandwidth. The maximum number of terminal stations (customer premises) is related to the LMDS base station capacity, as well as the data transmission rate required by each customer. In this case, the maximum range of the LMDS system is very important.

Modulation and channel bandwidth have a main impact onto the capacity of a LMDS and its range. The capacity depends on the spectral efficiency of used modulation schemes and the system range depends on the required E_b/N_0 at the receiver input for signal demodulation at a given bit error rate (BER). Some examples are presented in Table 17.11.

Spectrum efficiency not only depends on the used modulation method, but also on the channel filter mask roll-off factor, which is important to determine the correct relationship between channel bandwidth and data rates. For these reasons, real values of spectrum efficiency for LMDS systems differ from the theoretical and maximal ones, as presented in Table 17.11.

The system capacity increases for wider channel bandwidths, but the range decreases because of the higher noise level. The channel spacing in Europe is usually obtained by successive division of 112 MHz by 2. Table 17.12 contains detailed information about frequency bands preferred for LMDS systems, regulations and channel spacings. The channel arrangements and minimum PMP base station transmission capacity have been presented in Table 17.13.

The capacity in LMDS uplink and downlink channels usually differs because of physical layer function differences of both channels, even if the bandwidth is the same. In PMP systems physical

Table 17.11 Spectral efficiency and required E_b/N_0 for the modulations used in LMDS systems.

Modulation	QPSK	4QAM	16QAM	64QAM
Theoretical maximum spectral efficiency [bps/Hz]	2	4	6	8
Maximum spectral efficiency of LMDS systems [bps/Hz]	1.5	1.5	3.5	5
E_b/N_0 @ BER $= 10^{-6}$	10.5	15	18.5	24

Table 17.12 Frequency bands available to PP and PMP systems.

Band [GHz]	Frequency range [GHz]	Regulations and guidelines for channels arrangements	Channel spacing [MHz]
3.5	3.4–3.6	ERC/REC 14-03	1.75; 3.5; 7; 14
3.7	3.6–3.8	ERC/REC 12-08	1.75; 3.5; 7; 14
10.5	10.15–10.30 10.50–10.65	ERC/REC 12-05	3.5; 7; 14; 28
26	24.25–26.6	ITU-R F.748, ERC/REC 13-04; ERC/REC 13-02; ERC/REC 00-05	3.5; 7; 14; 28; 56; 112
28	27.5–29.5	ITU-R F.748, CEPT REC 13-04; 13-02; DEC (00)09	3.5; 7; 14; 28; 56; 112
32	31.8–33.4	ITU-R F.1571, ERC/REC 01-02	3.5; 7; 14; 28; 56
38	37–39.5	ERC/REC T/R 12-01	3.5; 7; 14; 28; 56; 140

Table 17.13 Channel arrangements and minimum PMP base station transmission capacity (based on [36,37]).

Minimum CRS transmission capacity [kbps] for modulation	Channel spacing [MHz]						
	1.75	3.5	7.0	14	28	56	128
4QAM	21 × 64	42 × 64	84 × 64 4 × 2048	8 × 2048	16 × 2048	32 × 2048	64 × 2048 or STM1
16QAM	42 × 64	84 × 64 4 × 2048	8 × 2048	16 × 2048	32 × 2048	64 × 2048 or STM1	128 × 2048 or 2 × STM1
64QAM	3 × 2048	6 × 2048	12 × 2048	24 × 2048	48 × 2048	96 × 2048 or STM1	192 × 2048 or 2 × STM1

layer and medium access control layer issues (i.e. channel coding, filtering, access and multiplexing methods) have to be taken into consideration during network planning. Each sector of PMP base station (CRS) shall provide minimum transmission capacity within the channel spacing. Requirements for each frequency band are defined in the appropriate standards (i.e. EN 301 213-2, EN 301 213-3 for systems in the frequency range 24.25 to 29.5 GHz which use FDMA and TDMA respectively).

The PMP system capacity can be considered as the maximum number of terminal stations simultaneously connected to the CRS sector and transporting their maximum payload bit rate. But the minimum payload (expressed either as the number of 64 kbit/s signals or an aggregate bit rate for the used traffic type, e.g. ATM cells) shall be taken into account as well. The maximum number of terminal stations simultaneously connected to the LMDS base station shall be declared by the manufacturer. That number can be calculated for each access method. PMP systems are limited by capacity and range. The network capacity can be increased through the use of sectorisation and frequency reuse mechanisms that allow the simultaneous use of the same frequencies in base station and the network. Typically, the LMDS base station provides service with 4, 8, 14, 16 or 24 sectors using directional antennas with 90, 45, 30, 22.5 or 15-degree beam-width respectively. Capacity can be increased by using different modulation schemes for different sectors. Some configuration examples of an LMDS base station are presented in Figure 17.29. The available configurations depend highly on the manufacturer's solutions.

The LMDS system coverage depends on several factors: frequency band, bandwidth, modulation type, type of the terminal being used, system availability and rainfall. The system availability is guaranteed by appropriate fading margins depending on frequency band and installation place, i.e. rain zone.

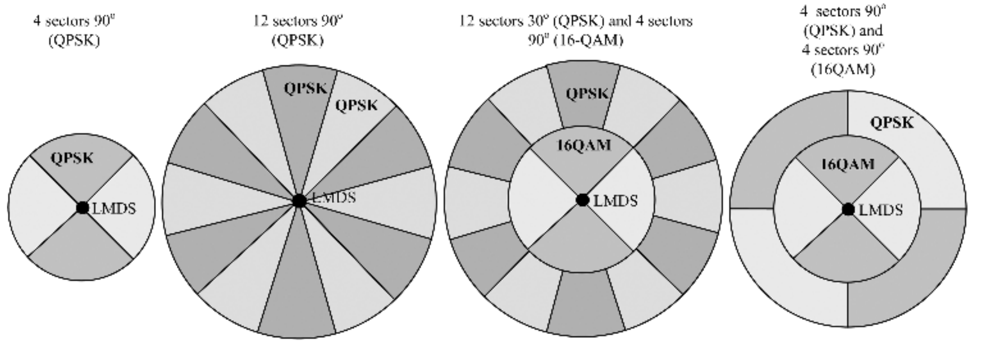


Figure 17.29 Examples of LMDS base station configurations.

The transmitter output power for CRS, RS and TS is limited to +35 dBm. For the same system capacity, the system coverage depends on transmitting and receiving antenna gains. Increasing the antenna gain of the terminal station, the service range can be expanded (Table 17.14). In this case, antennas with higher gains can be applied. But high gain antennas have to comply the standards (e.g. ETS 300 833). It must be noted that the LMDS station range can also be noticeably dependent on the used polarisation. The range difference for horizontal and vertical polarisations rises with frequency and depends on the ITU rain zone. Usually, the terminal station antennas are designed to be highly directional with high gain and narrow beamwidth (e.g. 18 dBi and 18° @ 3.5 GHz, 35 dBi and 2.5° @ 26 GHz).

Co-channel interference is caused by frequency reuse at terminal station receivers by remote LMDS base station transmitters and at LMDS base station receivers by remote terminal station transmitters. The interference imposed by TS transmitters on remote LMDS BS receivers is more detrimental than the interference imposed by LMDS BS transmitters onto remote TS receivers, because the former hinders reliable reception for all terminal stations served from the remote BS, while the latter only affects a few TSs.

The co-channel and adjacent-channel interferences can be reduced by physically separating potential interferers from each other and maximising the isolation between adjacent sectors in the network through orthogonal polarisation (horizontal and vertical) [39]. The antennas with narrow beamwidths located as high above ground as possible allow reducing multipath propagation effects, cross-polarisation and co-channel interferences in the network.

The lower the interference levels, the higher the frequency reuse factor and the network capacity for the same radio resources. Higher reuse factors reduce interference more than lower reuse factors, but not without cost; for a fixed total spectral allocation, higher reuse factors reduce the amount of usable spectrum per LMDS base station. Conversely, to maintain the same amount of usable spectrum per LMDS BS, higher reuse factors require larger total allocations. A smaller allocation requires a denser grid and more LMDS base stations, increasing costs (also backbone and switching). Conversely, a larger allocation allows a sparser grid with fewer LMDS base stations and lower costs. Some frequency allocation models for LMDS networks, which also include polarisation separation, are presented in Figure 17.30.

17.4.3.4 Local Multipoint Distribution Systems (LMDS) Solutions

The LMDS can support a variety of services over the same infrastructure and a wide range of customers can be served from the same base station. For these reasons, it is the perfect *last mile* solution for many communication and information technologies: IP, ATM, Ethernet, Frame Relay, Leased Line,

Table 17.14 Examples of LMDS range in kilometres.

Frequency band [GHz]	Modulation schemes	Base station sector [°]	Terminal antenna	Rain zone E				Rain zone H			
				7 MHz channel		14 MHz channel		7 MHz channel		14 MHz channel	
				Availability				Availability			
				99.999 %	99.99 %	99.999 %	99.99 %	99.999 %	99.99 %	99.999 %	99.99 %
10.5	QPSK	90	integrated	13.7	21.4	10.8	16.7	9.9	16.3	8.2	13.3
	16-QAM	90	integrated	9.6	15.0	7.6	11.7	6.9	11.4	5.7	9.3
26	QPSK	30	integrated	3.1	4.9	2.8	4.3	2.5	4.2	2.2	3.6
			390 mm	4.1	6.6	3.7	5.8	3.3	5.7	2.9	5.0
			650 mm	5.1	8.4	4.6	7.5	4.0	7.3	3.6	6.5
		90	integrated	2.7	4.3	2.4	3.7	2.2	3.7	1.9	3.2
			390 mm	3.6	5.7	3.2	5.1	2.8	5.0	2.5	4.4
			650 mm	4.4	7.3	4.0	6.5	3.5	6.4	3.1	5.6
	16-QAM	30	integrated	2.2	3.4	1.9	3.0	1.8	4.2	1.6	2.5
			390 mm	2.9	4.6	2.6	4.1	2.3	2.9	2.1	3.5
			650 mm	3.6	5.9	3.2	5.2	2.8	5.1	2.5	4.5
		90	integrated	1.9	3.0	1.7	2.6	1.5	3.7	1.4	2.2
			390 mm	2.5	4.0	2.2	3.5	2.0	2.6	1.8	3.1
			650 mm	3.1	5.1	2.8	4.5	2.4	4.5	2.2	4.0

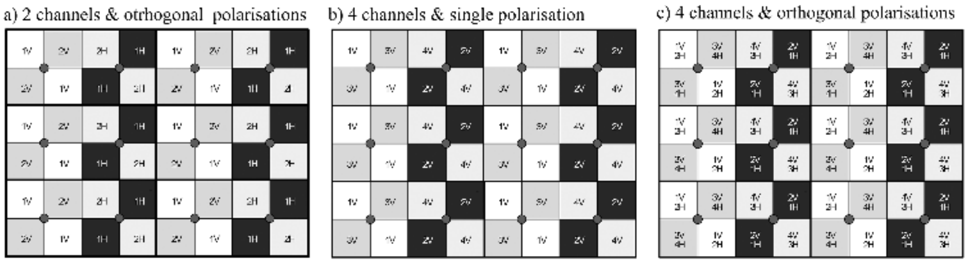


Figure 17.30 Frequency allocation models for LMDS network.

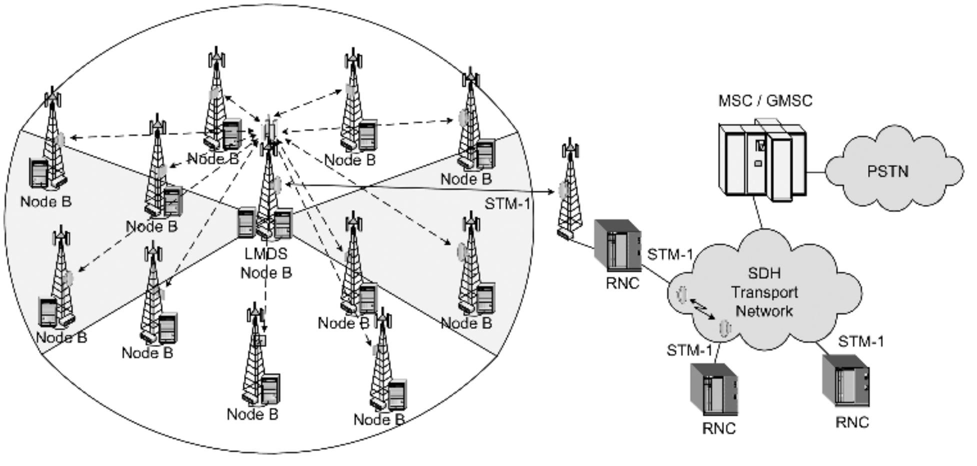


Figure 17.31 An example of LMDS backhaul network for UMTS.

POTS and ISDN, which provide end users different services: VoIP, data transmission and access to the Internet. It is a flexible solution for interconnecting 2G, 3G and collocated 2G/3G base stations to the transmission network (Figure 17.31). The LMDS system provides telecommunication services to small and medium enterprises (SME), small offices/home offices (SOHO) and residential customers, and ensures wireless local loop backhaul. The modular and scalable system architecture allows the operator start deploying small networks with low CAPEX and low risk and expands the network if necessary. Additionally, LMDS and PMP systems enable efficient and fast installation of new elements (terminal stations), re-deployment and network modifications according to new operator demands. Dynamic bandwidth allocation and control allow an efficient use of system resources. Efficient backhaul connectivity provides coverage for mobile and fixed narrowband wireless system base stations with cost effective transmission infrastructures. In the last several months, operators have become interested in the new WMAN systems, which offer much higher transmission rates then LMDS systems without LOS requirement. In the future WMAN systems will replace LMDS systems.

17.4.4 WiMAX AS A POTENTIAL UTRAN BACKHAUL SOLUTION

The system, which recently gained big market popularity, is WiMAX. Because its main application deals with point-to-multipoint broadband wireless access, it was natural to assess its UTRAN backhauling possibilities. The following section gives a WiMAX system overview, whereas more practical guidelines can be found in Section 17.5.

17.4.4.1 WiMAX Standard Overview

As already elaborated on in Section 4.5.2, the WiMAX system concept originated from the IEEE 802.16 standard family; this relation is very similar to the one between WiFi and the IEEE 802.11 standards. The entire IEEE 802.16 group is designed for carrier class metropolitan access. The family consists of the following standards:

- IEEE 802.16 – main Fixed Wireless Access standard;
- IEEE 802.16a – enhancement for 2–11 GHz band;
- IEEE 802.16b – Quality of Service;
- IEEE 802.16c – enhancement for 10–66 GHz profiles (interoperability);
- IEEE 802.16d – adapts 802.16a to mobility requirements;
- IEEE 802.16e – completes support for mobility (including handover);
- IEEE 802.16f/g – network management extensions;
- IEEE 802.16h – license exempt operation.

The main standard in the family is IEEE 802.16 [40]. It specifies the radio interface for point-to-multipoint broadband wireless access systems working on frequencies from 10 to 66 GHz. Medium Access Control (MAC) layer and three physical layers (OFDM, OFDMA and single carrier) are specified. The most popular standard currently is called 802.16 Revision D (or 802.16-2004). It was approved in July 2004 and focuses on fixed applications and smart antenna enhancements for indoor applications. It also extends the system band to frequencies below 11 GHz. It consolidates the base standard and all the amendments (‘a’ and ‘c’). Therefore, most of the WiMAX equipment currently available on the market is compliant with the Revision D standard.

The future standard which specifies Mobile WiMAX is 802.16e. It incorporates features and protocols needed for portability and mobility. It also includes some new modes, such as SOFDMA, to enhance the portability and mobility performance. Commercial applications of this standard are expected in 2007–2008.

In parallel to the IEEE effort to standardise WiMAX, there are also related works in ETSI, dealing with the HiperMAN standard. The European standard is in fact identical to the IEEE 802.16 Revision D, except that the only supported physical layer is OFDM.

The third important body for the system standardisation is the WiMAX Forum. It is an industry organisation appointed for a worldwide promotion of IEEE 802.16 and ETSI’s HiperMAN standards. The main goal was to have one choice for physical layer and optional parameters, to enable interoperability and industry usage of the common technology. The WiMAX Forum also drives the effort for test specifications and performs test lab equipment certification processes.

It is also worth mentioning that WiMAX is under evaluation as a future potential UMTS access method (802.16e standard) in the framework of 3GPP standardisation. More information on WIMAX standards can be found on [41,42].

17.4.4.2 System architecture

The typical WiMAX system architecture as depicted in Figure 17.32 consists of:

- Base Station (BS);
- Customer Premises Equipment (CPE);
- Repeater;
- Network Management System.

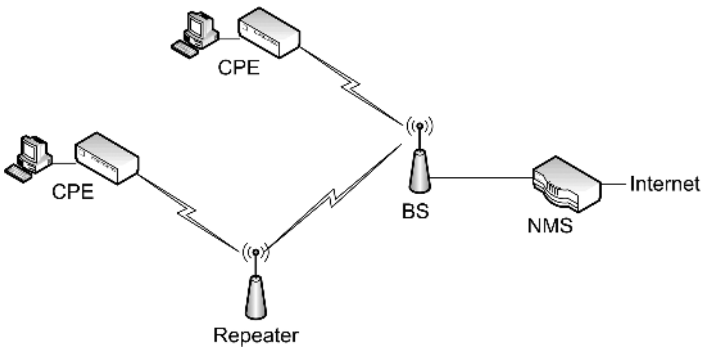


Figure 17.32 WiMAX System architecture.

The typical base station configuration is from one (omnidirectional) up to six sectors. The repeater is usually similar to the base station, but without backhaul connectivity; this enables range extension. There can be two main types of CPE: outdoor and indoor. While using outdoor ones, the same system coverage can be achieved with a less number of base stations (only outdoor coverage service is needed). The role of NMS is WiMAX network management. Additionally, the system should have some routers and switches for traffic connectivity and aggregation.

17.4.4.3 System Characteristics

The WiMAX system, compatible with IEEE 802.16 Revision D standard, can yield up to 70 Mbps in a 20 MHz channel in the frequency range from 2–11 GHz. However, the equipment available now uses mainly 3.5 MHz channels that can achieve 12.7 Mbps throughput in one channel. The second currently available channel size is 1.75 MHz. The other possible channel sizes are: 5, 7, 10 and 20 MHz. The system can use both FDD and TDD duplexing methods, which depends on the frequency band used. The usual FDD duplex separation is 100 MHz. The bands available for WiMAX are:

- 3.4–3.6 GHz (currently the most popular band; 1.75 MHz, 3.5 MHz channels in FDD mode available now; 7 MHz channel in TDD mode available in future);
- 2.5–2.7 GHz (5 MHz channels in TDD mode);
- 3.6–3.8 GHz (possible extension band for 3.4–3.6 GHz);
- 5.1–5.3 GHz (possible future extension band);
- 5.4–5.8 GHz (5 and 10 MHz channel in TDD mode; license exempt band);
- 698–746 MHz (possible future extension band).

The only available physical layer is OFDM with 256 sub-carriers (FFT size). The modulations and code rates supported are:

- BPSK (1/2, 3/4);
- QPSK (1/2, 2/3, 3/4);
- 16QAM (1/2, 3/4);
- 64QAM (2/3, 3/4, 5/6).

In fact, the system is using an adaptive modulation scheme. Therefore, the optimal modulation is chosen in dependency of the particular environment scenario. Table 17.15 presents example throughputs that can be achieved on a single channel.

Table 17.15 Example WiMAX single channel throughput.

Modulation	Channel bandwidth	
	3.5 MHz	1.75 MHz
BPSK 1/2	1.41 Mbps	0.71 Mbps
BPSK 3/4	2.12 Mbps	1.06 Mbps
QPSK 1/2	2.82 Mbps	1.41 Mbps
QPSK 3/4	4.23 Mbps	2.12 Mbps
QAM16 1/2	5.64 Mbps	2.82 Mbps
QAM16 3/4	8.47 Mbps	4.24 Mbps
QAM64 2/3	11.29 Mbps	5.56 Mbps
QAM64 3/4	12.71 Mbps	6.35 Mbps

The other important system features are protocol independent core and QoS mechanisms, as well as security. From the core perspective, it is important that WiMAX can transport IPv4, IPv6, Ethernet, ATM or others, supporting multiple services simultaneously. The standard which deals with QoS is 802.16b. It enables non line-of-sight (NLOS) operation without severe distortion of the signal from buildings, weather and vehicles. It also supports intelligent prioritisation of different forms of traffic according to its urgency. The security mechanism provided in WiMAX includes measures for privacy and encryption, such as authentication with x.509 certificates and data encryption using DES in cipher block chaining (CBC) mode with hooks defined for stronger algorithms like Advance Encryption Standard (AES).

17.4.4.4 Examples of System Applications

According to the WiMAX Forum vision, the goal for the system is to create a global mass market for deployment of broadband wireless networks, which will enable fix, portable and mobile users to maintain high speed connectivity and to lead the 'access everywhere' revolution supporting delivery of data, voice and video applications at home, in the office and on the go. Thus, and because of wide features and functionalities of WiMAX, the system can address different market segments:

- Business access market;
- Residential broadband access (Internet + other services);
- Cellular network backhaul;
- WiFi hotspots connectivity.

For business users (enterprises), WiMAX provides an equivalent of N times E1/T1, ATM, Frame Relay or Fast Ethernet connectivity and fractions of them. The system can be used instead of fibre networks or xDSL connections with greater cost effectiveness for subscribers. Furthermore, for residential access the system is either equivalent of xDSL and cable TV access. Especially, while designing the network for outdoor CPE usage, the cost effectiveness seems to be far better than for classical copper line systems.

The main WiMAX usage, from this chapter's perspective, are cellular networks (including UMTS) backhauling. The details on backhauling possibilities and limitations of PMP system are presented in Section 17.6. The subset of this, but with slightly different demands, is WiFi hotspot connectivity; particularly for cities with a large density of WiFi hotspots it provides more cost effective solutions for WiFi hotspot transport.

17.4.4.5 Possible Evolution Paths

The main WiMAX evolution path paves the way to mobility dimensioning. As current WiMAX market implementations may be named fixed outdoor systems, the fixed indoor and nomadic ones are only one step ahead. All of these are connected with the 802.16 Revision D standard, and the main differentiating factor for them is the CPE type (outdoor or indoor). Furthermore, fixed outdoor requires fixed location, installation outside the subscriber's house and provides applications of E1/T1 or fractional E1/T1 level services for enterprises, backhauling solutions and limited broadband access (early adopters, rural and developing countries). Whereas fixed indoor enables consumer self installing and auto provisioning, gives possibility of nomadism (subscriber can move CPE to another location in service area) and provides broader range of last mile broadband access for residential consumers.

Further evolutionary steps are related to the 802.16e standard; they are hence about portability and mobility. The portable version is dependant on WiMAX enabled chipsets incorporated into laptop PCs (plans have already been announced by Intel). It will provide some handover mechanisms, but with 1–2 second interruption type. The seamless handover for real-time services will be available in the full mobile version only. The devices will be PDA or mobile phone types, rather than mobile PCs. As mentioned before, this version is expected to be incorporated into the 3GPP standards framework, as a complementary UMTS access method.

17.5 EFFICIENT USE OF WIMAX IN UTRAN

WiMAX systems seem to be a big opportunity for usage as UTRAN complementary backhaul solutions. For the UMTS system planner, every possibility that can optimise investments should be evaluated. Therefore, it is very important to assess CAPEX and OPEX savings that can incur because of a WiMAX implementation and to estimate all the technical drawbacks and limitations of such solution. This section deals with both of them.

17.5.1 DIMENSIONING OF WIMAX FOR UTRAN INFRASTRUCTURE

To assess the reasonability of using WiMAX for UTRAN backhauling, a similar method as described in Chapter 8 should be used. All the system implementation and maintenance costs should be calculated and compared with alternative solutions (e.g. LMDS, point-to-point microwave links, leased lines). Since the results can be different per clutter type, the case that suits a particular environment best should be used for comparison purposes. Probably the best method is to compare live transmission network costs in a particular area with the costs of hypothetical WiMAX rollout. To enable such a comparison, some initial WiMAX network assumptions should be made, e.g. Node B collocated with 2G BTS, WiMAX BS not collocated with 2G BSC, connection between Node B and RNC via WiMAX and ATM network, WiMAX system equipped with E1 interfaces, etc.

Given the assumptions, both coverage and capacity dimensioning phases should be performed. First of all, the WiMAX BS configuration should be decided upon. An example BS configuration could be:

- 4 sectors;
- 7 MHz of bandwidth per sector;
- resulting average capacity of 14 Mbps per sector.

All the other BS physical parameters should be properly set as well, in order to enable power budget construction and pathloss (cell range) calculation. For ease of comparison, it is useful to convert the total site throughput to E1s. By dividing the particular network area by the WiMAX coverage (cell) area, the number of WiMAX sites is obtained. Furthermore, having the sum of all 3G sites' transmission needs (in E1s) divided by the WiMAX BS capacity, the number of WiMAX capacity sites is calculated. The bigger of these values is the number of WiMAX sites needed.

The WiMAX CAPEX consists of the number of sites multiplied by the single site cost (equipment, civil work, etc.) plus the WiMAX license cost (it is important to mention that, across most countries, the WiMAX license can be nationwide or assigned for a particular administrative area only). The WiMAX OPEX consists mainly of equipment maintenance that can be easily calculated with drivers related to CAPEX (as presented in Chapter 8).

While considering a completely new network and CAPEX related solutions (microwaves or LMDS), a credible financial comparison with other backhauling methods can be done by simply comparing related CAPEXs. When comparing WiMAX versus leased lines, the sums of cumulative CAPEX plus OPEX should be compared. In this case, the advantage of WiMAX is only the matter of time (thus, also the PayBack Time can be calculated). Furthermore, to evaluate the savings because of a possible migration of existing backhauling technologies to WiMAX, similar methods can be used, but with serious attention paid to PayBack Time.

The results of such business case scenarios performed by the authors clearly showed WiMAX to be applicable in relatively dense and clustered towns. While comparing to leased lines (according to current pricings), a WiMAX solution is always cheaper (different PayBack Times). And finally, a WiMAX solution is also cheaper than microwave links when one WiMAX BS can support more than 10–12 Node Bs.

17.5.2 CURRENT WIMAX LIMITATIONS

From the previous section, it can be incurred that WiMAX could be a very good and cost effective UTRAN backhauling technology, at least in particular scenarios. However, looking into deep technical details, some serious limitations may also be noticed. One of them is the total WiMAX network capacity limit, which is not sufficient for future UTRAN capacity needs. The other one is the network synchronisation issue.

The mobile network radio interface requires a synchronisation stability of $5 \cdot 10^{-8}$ s [43–45]. Such stability cannot currently be provided while connecting Node Bs via a packet switched network (like WiMAX). It hence seems that synchronisation can be a serious blocking factor for using WiMAX as a UTRAN backhauling system. Fortunately, there are three candidate solutions that can overcome this issue. These are:

1. Network Time Protocol (NTP);
2. IEEE 1588;
3. SYNCoIP.

The already established protocol and readily available solution at the server side is a local reference steered by the NTP. However, the possibility to achieve the required accuracy is uncertain. Furthermore, because of the long time constant, the cost of a local reference is relatively high.

The next possible solution for synchronisation over packet is IEEE 1588. With this standard, the reference is maintained and distributed through the Ethernet network. It is an already standardised protocol, which has a more than required accuracy. It could also easily be integrated into WiMAX terminals. But it requires an Ethernet based network only and must be supported at each network element; this is clearly more difficult, because the protocol is relatively new, with not so many commercial implementations.

The last way to overcome the synchronisation issue is SYNCoIP: a reference is transported over IP in ‘CE style’. The drawback of that method is that the development is not finalised and IP QoS support may be needed for that method to work. On the other hand, it provides the required accuracy, relies on IP only and is developed from existing techniques.

Concluding, the synchronisation issue seems to be a hard blocking factor for WiMAX implementation as a UTRAN backhauling solution now. Current methods to overcome this issue are either not widely available on the market or not even mature enough for implementation. However, there are real chances for having synchronisation over WiMAX in the very near future.

17.6 COST-EFFECTIVE RADIO SOLUTION FOR UTRAN INFRASTRUCTURE

In the previous sections, it has been shown that the UMTS network imposes some strict demands on the UTRAN transmission infrastructure. These demands, in comparison to other systems as e.g. GSM, arise from the higher capacity of UMTS networks, especially in dense urban areas where a significant increase in the number of Node Bs and a growth of the bit rate are being observed. Additionally, the number and type of served services as well as transport format (ATM or IP) effect the capacity of UTRAN transmission links. In many cases, the transmission infrastructure has to serve base stations of two networks (GSM and UMTS), and requirements for both systems have to be considered during backhaul network planning.

The UTRAN backhaul network has to fulfil high transmission requirements of Node Bs and provide a low purchase and operating cost, good transmission quality and reliability, flexibility and scalability. Some possible solutions for UTRAN transmission infrastructure have been presented in Sections 17.4 and 17.5. Due to an easy network installation and extension, the fixed radio systems (e.g. PTP, PMP, LMDS and WiMAX) seem to be the most suitable. For the PTP systems, the spectrum efficiency is the highest, but the costs, large number of required antennas and inflexibility of this approach rise serious problems. The PMP or WiMAX systems are much more flexible, but their cost-effectiveness depends on the application. In the case of supporting a UMTS backhaul network, dedicated methods of finding a cost-effective solution are required. These methods and some comparative results for applications of PTP and LMDS in UTRAN will be presented in following sections.

17.6.1 RF PLANNING ASPECTS

Radio backhaul network planning for UMTS covers a series of problems related to both UMTS and transmission system planning. Before commencing with the design of the backhaul network, which is to serve the UMTS system, it is necessary to define the key parameters of UMTS. These parameters affect the architecture and parameters of the UTRAN transmission network. The most important for the analysis are UMTS base station locations and required throughput according to service types and transmission overhead. All this information can be derived from the real UMTS network or any suitable planning tool. Having the UMTS environment defined, the radio backhaul network can be introduced.

Correct RF engineering and planning of the backhaul network requires ensuring optimal deployment of the network. The output of these tightly coupled engineering exercises results in a multi-year plan for the deployment and installation of radio systems (PMP and PTP) and its backbone transport and switching infrastructure. The goal in all RF engineering and backhaul network planning aspects is to provide high-quality services over high-availability wireless links to as many Node Bs as possible within each localised region in a resource-efficient manner. This means maximising the RF range, taking into consideration the statistical nature of the radio performance. The radio system specifications (gains, transmit powers, receiver sensitivities, etc.), when matched to worst-case rainfall and atmospheric attenuations and interference-induced degradations, yield the radio system range of the fixed radio system; the latter is defined by an operator at a minimum required BER and a percentage of the time the network can provide services. For backhauling applications, the operator shall demand a BER of at least 10^{-6} and a system availability of 99.99 or 99.999 %; poor service quality or system availability of the radio backhaul network can damage business viability of the UMTS network.

It has to be noted that parameters and functionality of fixed radio transmission systems differ from vendor to vendor; therefore, ranges shall be considered for each system installation independently. In particular, it concerns the rain region, used frequency band, channel spacing, modulation scheme, terminal types and antenna gains. The maximum service radius is an important design parameter of the backhaul network. It mainly concerns systems with a central base station (PMP, LMDS and WiMAX), because the number of base stations required for a complete coverage is inversely proportional to the maximum cell radius of each of them. Usually, manufacturers publish only maximum idealised interference-free service ranges for some typical configurations. The service range examples of LMDS for different configurations, consisting of 90- and 30-degree sectors and three different subscriber type antennas, have been presented in Table 17.14.

Frequency reuse is necessary in the real backhaul network with many base stations; however, this causes interferences at some locations within the service area. These interferences can degrade radio transmission performance (range, throughput, availability and quality) of some radio links and for this reason shall be taken into account for planning complex and wide networks to reach required transmission performance to all Node Bs in the UMTS network.

For all radio systems (PTP, PMP, LMDS, WiMAX), the best transmission performance is achieved for LOS conditions with the first Fresnel zone devoid of any obstacles. Only WiMAX systems can operate without direct line of sight, but in this case their ranges and availability decrease dramatically. For this reason, during UMTS radio planning, installation of additional radio equipment for a backhaul network requiring LOS conditions to other devices shall be considered. The only practical solution is the use of sophisticated planning procedures, implemented in planning tool. These procedures should study inter-PMP interferences and PTP-on-PMP interferences; they should also be able to find the best Node Bs locations according to UMTS *and* the backhaul network performance. The choice of a solution for the UTRAN transmission infrastructure depends on the radio planning results and the cost analysis.

17.6.2 THROUGHPUT DIMENSIONING

A very important part of designing the UTRAN transmission network is the dimensioning of the required throughput for each Node B in the UMTS network. For this reason, the output of radio network planning is taken as input for the radio access network design. All the data can be divided into a number of parts, which represent clusters in RNC service areas. For each cluster, the number of base stations, their location and the maximum traffic demand (including all system procedures and services) have to be known. The calculation of maximum traffic demands shall consider not only traffic in the UMTS radio interface, but also control information indispensable for managing Node Bs. The transmission formats used in the interface between Node B and RNC in conjunction with served services are of great importance to the calculations to the Node B's throughput. It is mainly related to the transmission overhead, which varies from service to service. Detailed requirements for UTRAN transmission infrastructure and throughput dimensioning have been presented in Sections 17.1–17.3.

The accurate analysis of maximum Node B throughput requirements, information about distance and propagation conditions in the radio path to each other Node B in the cluster are sufficient to design the UTRAN transmission infrastructure using Point-to-Point radio systems. Some types of radio lines used in backhaul networks and their parameters have been presented in Table 17.16.

For the backhaul radio network solution, which makes use of the central station, the designer has to know the total capacity of all Node Bs in the cluster and the cluster size, which is defined as the maximum distance from between remote Node Bs in the cluster. To design the radio backhaul network, the operator has only to know the radio resources, which are being assigned to the operator by the

Table 17.16 Parameters of radio relay systems used in UTRAN backhaul networks.

Type	No. of E1 interfaces	Total throughput (kbps)	Channel spacing (MHz)
PDH 2E1	2	4096	3.5
PDH 4E1	4	8192	7
PDH 8E1	8	16384	14
PDH 16E1	16	32768	28

national spectrum regulator. The amount and bandwidth in assigned channels, as well as frequency band and Node B layout, determine the central station configuration, service area range and number of base stations in the cluster. Of course, during the designing process, the operator has to take into consideration all the aspects mentioned in this chapter.

Because analysis and design are quite similar for PMP, LMDS and WiMAX systems, only the LMDS solutions will be considered here. The LMDS base station parameters and configuration have a vital impact onto the system capacity. On the other hand, they highly depend on the actual system being used. For the purpose of analysis, let us consider the potential capacity of modern commercial LMDS systems from a major manufacturer. The LMDS central station (CS) capacities for some configurations and assigned radio resources with 7 MHz channel spacing have been presented in Table 17.17.

All considered configurations provide the co-channel and adjacent-channel interference reduction by using other radio channels in adjacent sectors for QPSK and 16QAM modulation. For this reason, LMDS base station configurations with QPSK and 16QAM require at least four radio channels. This assumption does not allow configuring LMDS base stations if the spectrum regulator has assigned the operator only two radio channels. It is theoretically possible to assign the same channels to all sectors (QPSK or 16QAM), but the orthogonal polarisations in the adjacent sectors have to be used. The transmission with 16QAM modulation is more susceptible to interference than QPSK, given its higher carrier-to-noise requirements; therefore, one-frequency solutions with 16QAM LMDS base stations are not recommended.

Regardless of the frequency band, the sector capacity only depends on the modulation scheme, channel number and channel spacing. For this system, the useful net bit rate in the 7 MHz channel with QPSK modulation is 8.192 Mbps; however, due to the required overhead (forward error correction, encryption, MAC messages) the raw bit rate is about 12 Mbps. The sector capacity doubles if 16QAM modulation is implemented instead of QPSK, or if 14 MHz channels are used instead of 7 MHz channel.

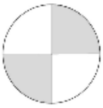
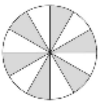
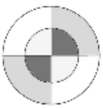
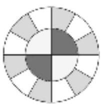
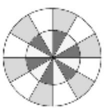
Denser sectorisations provide higher spectrum efficiency, higher LMDS base station capacity and larger ranges. However, this should only be done for a high Node B density in the LMDS coverage area, otherwise it is possible that some too narrow sectors will not serve any Node B.

17.6.3 METHODS OF FINDING OPTIMAL LMDS NETWORK CONFIGURATIONS

Optimal network configuration means that the UTRAN transmission infrastructure will provide required capacity and will operate with a minimum number of equipment and radio resources. To solve this problem, a suitable optimisation method is necessary; however, there is no single method available, which allows finding the optimal radio backhaul network solution for UTRAN. Due to the many possible solutions, we limit our considerations only to some methods of finding the optimal LMDS network configuration. All these methods can be adapted to any other backhaul network solution, which makes use of radio systems with a central base station, e.g. PMP or WiMAX.

The definition of optimal network configuration has to be modified for LMDS used for the UTRAN transmission infrastructure. In this case, the LMDS base station shall serve all Node Bs in the coverage

Table 17.17 LMDS central station capacity for 7 MHz channel spacing.

Configuration	Radio resources	14 MHz	28 MHz	56 MHz	112 MHz
	Modulation	QPSK	QPSK	QPSK	QPSK
	Channels in sector	1	2	4	8
	Sector capacity Mbps	8	16	32	64
	Total CS capacity Mbps	32	64	128	256
	Channels in CS	4	8	16	32
	Modulation	QPSK	QPSK	QPSK	QPSK
	Channels in sector	1	2	4	8
	Sector capacity Mbps	8	16	32	64
	Total CS capacity Mbps	96	192	384	768
	Channels in CS	12	24	48	96
	Modulation	—	QPSK+16QAM	QPSK+16QAM	QPSK+16QAM
	Channels in sector		1 1	2 2	4 4
	Sector capacity Mbps		8 16	16 32	32 64
	Total CS capacity Mbps		192	384	768
	Channels in CS		8	16	32
	Modulation	—	QPSK+16QAM	QPSK+16QAM	QPSK+16QAM
	Channels in sector		1 1	2 2	4 4
	Sector capacity Mbps		8 16	16 32	32 64
	Total CS capacity Mbps		160	320	640
	Channels in CS		16	32	64
	Modulation	—	QPSK+16QAM	QPSK+16QAM	QPSK+16QAM
	Channels in sector		1 1	2 2	4 4
	Sector capacity Mbps		8 16	16 32	32 64
	Total CS capacity Mbps		288	576	1152
	Channels in CS		24	48	96

area with a transmission quality and availability required by the operator, or better. It shall limit all implementation and operational costs of the LMDS network and provide an efficient utilisation of all the equipment as well as allocated spectrum.

All the methods can be divided into two groups. In the first one, the analysis is performed only for a theoretical deployment of UMTS base stations according to the basic cellular system design methods with regular and triangular grid and hexagonal cells. The range and capacity of each Node B are the same for the entire network. The UMTS base station range depends only on the link budget calculated for the given transmitter and receiver parameters, environment, service types, number of frequency channels being used, loading, sector types and cell loading. The capacity is calculated for a given service or mix of services, required transmission overhead and control data. In the second type of methods, the analysis is preformed for a real UMTS network or for data obtained from UMTS radio network-planning tools. In this case, information about the Node B locations as well as their transmission link capacity is only required.

17.6.3.1 Approximation Method

To check the optimal LMDS network configuration, the total capacity of all Node Bs located in the LMDS serving area has to be compared to the total LMDS base station capacity. The single UMTS base station capacity and a number of such base stations in the LMDS coverage are solely necessary for this kind of calculation, given that a uniform UMTS network with regular cell range is considered. Due to many available LMDS configurations, the most convenient way is to do the analysis for a single LMDS sector and then multiply the obtained result by the number of sectors. For uniform UMTS networks, the function which defines the number of Node Bs in a single sector of an LMDS base station can be approximated by a polynomial given in [46]. The method is universal given that the approximation is applied to the relation between LMDS range and Node B radius. The number of Node Bs in an LMDS sector depends on the mutual location between LMDS and UMTS base stations; therefore, the approximation has to be prepared separately for each LMDS sector. There are two main LMDS base station locations with reference to the UMTS Node B. The first one is called *collocation* (Figure 17.33a), because the LMDS base station is placed together with one of the Node Bs. In the second one, which is called *no collocation*, the operator has to find an independent location for the LMDS base station (Figure 17.33b).

Knowing both system ranges and the polynomial function, it is possible to calculate the number of Node Bs in the LMDS sector and the total transmission capacity for a given service profile. The comparison of the maximum capacity offered in the LMDS sector and the required one by the UMTS base station located in it yields some information about efficiency. If the transmission capacity offered in LMDS sector is much higher than required by UMTS, this means that the LMDS range is limited and the LMDS base station serves too few UMTS Node Bs. Such a situation occurs usually in rural areas, where PTP solutions are much cheaper than LMDS ones. If the transmission capacity offered in the LMDS sector is not sufficient for UMTS requirements, this means that the LMDS capacity is limited and only some but not all Node Bs in the sector are served by the LMDS base station.

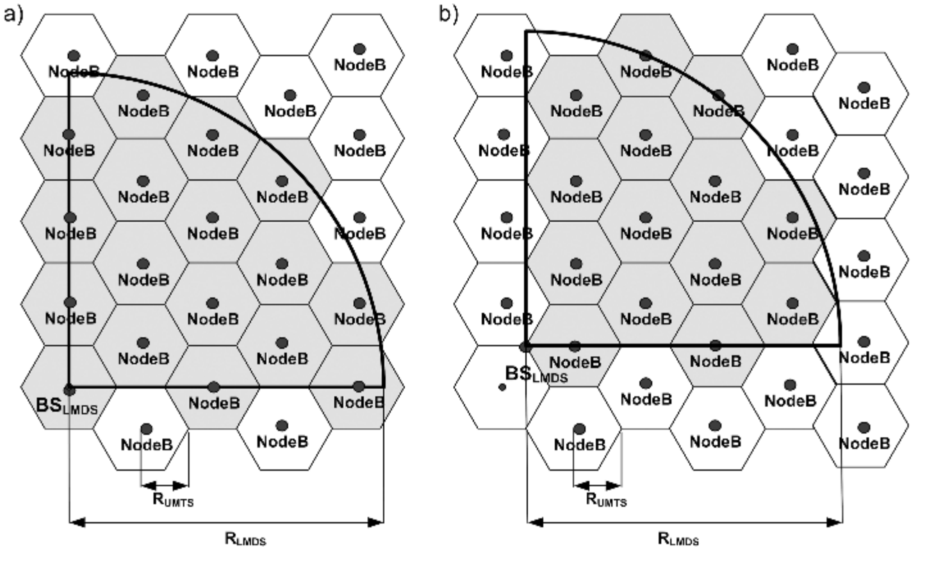


Figure 17.33 Possible LMDS base station location in reference to UMTS Node B: (a) collocation with Node B and (b) no collocation.

Thus, some efficiency factor of applying LMDS for UTRAN's backhaul network has to be known to find the optimal solution. The knowledge of the maximum LMDS sector range, which allows as many as possible Node Bs located in its coverage area to be served, is very important for planning purposes. This factor is referred to as the Maximum Sector Capacity Limited Range (M-SCLR). It has to be noted that the M-SCLR depends on the LMDS sector capacity, as well as on the Node B density and transmission requirements.

When comparing the range of a real LMDS sector with the calculated SCLR, it is possible to estimate its usefulness for the UTRAN backhaul network. If the range of a real LMDS system is the same or imperceptibly smaller then the maximum sector capacity range, this means that the LMDS configuration is (close to) optimum for the UTRAN transmission infrastructure. The LMDS sector capacity has to decrease, if the SCLR is greater than the real LMDS, because the LMDS configuration has likely been chosen wrongly (for example: too many sectors, or too many channels used in each sector, or channel bandwidth too broad). The LMDS capacity ought to be increased or the LMDS range decreased, if the sector capacity range is smaller than the real range.

In Table 17.18, some examples of the maximum sector capacity limited range for a 90-degree LMDS system and a uniform (theoretical) UMTS network with a loading of 25 % are presented. The sign '>>' in Table 17.18 means that the maximum sector capacity limited range of LMDS is much bigger than that for the longest LMDS range in the 10 GHz frequency band. The UMTS network design has been developed using the usual power budget and the COST-Hata propagation model. The Node B's maximum indoor range and required capacity to the RNC have only been determined for voice services (12.2 kbps) to UMTS low transmit power terminals (21 dBm). The Node B capacity was calculated for three sectors, taking into consideration the transmission overhead in ATM. Comparing this data to real LMDS ranges, it is possible to estimate whether the LMDS sector ranges are capacity limited or not. Such an analysis has been carried out for some of the (theoretical) UMTS networks serving only one type of service.

Calculation results for 90-degree LMDS sectors using a single channel of 14 MHz bandwidth for data transmission to and from UMTS base stations are presented in Table 17.19. In the frequency band of 26 GHz, the LMDS ranges do not exceed 7 km; thus, only for a few cases, an LMDS capacity limitation has occurred. This is usually observed for high bit rate services in dense urban environments, where QPSK is used; italic/bold fonts indicate all these cases. In other cases, the Maximum Sector Capacity Limited Range has exceeded the real range, and only the real range is written down in the table. Comparing these results to Table 17.18, it is clear that for a 16QAM scheme, the range is too short and capacity too high.

17.6.3.2 System Transmission Density

There is another way than presented in Section 17.6.3.1 to find an optimal LMDS base station configuration for UTRAN transmission networks. To determine the maximum LMDS sector capacity limited range for UTRAN backhaul networks, a comparison of the transmission density ratio for Node Bs and LMDS sectors versus the LMDS sector range has to be done [47]. The transmission density ratio is defined between the radio system transmission capacity (in bps) and the related surface of the served area (in m^2). The transmission density is hence expressed in $b/(s \cdot m^2)$ and combines two of the most important parameters. The Node B's transmission density function is calculated as the quotient of the Node B's required link throughput to the RNC and the surface of the coverage area. The LMDS sector transmission density is determined from dividing the LMDS sector throughput by its serving area.

The Node B transmission density can be assumed constant for specified environments and services. Additionally, it is independent from the LMDS sector range and distinct from the LMDS sector transmission density, which decreases with an increase of the LMDS sector range, achieving minimum value for maximum range.

Table 17.18 Maximum Sector Capacity Limited Range for LMDS 90° sector.

Environment				Maximum Sector Capacity Limited Range in km							
				Dense urban				Suburban			
Modulation scheme	LMDS channel bandwidth	Single LMDS channel throughput	Number of channels assigned to single sector	Voice 12.2 kbps	LCD64 kbps	LCD144 kbps	LC384 kbps	Voice 12.2 kbps	LCD64 kbps	LCD144 kbps	LC384 kbps
QPSK	7 MHz	8 Mbps	1	3.00	3.40	2.75	2.10	6.95	7.95	2.75	4.90
			2	4.45	5.10	4.15	3.05	10.35	11.85	4.15	7.10
			3	5.80	6.40	5.20	3.80	13.45	14.75	5.20	8.80
			4	6.70	7.45	6.05	4.40	15.50	17.20	6.05	10.20
	14 MHz	16 Mbps	1	4.45	5.10	4.15	3.05	10.35	11.85	4.15	7.10
			2	6.70	7.45	6.05	4.40	15.50	≫	6.05	10.20
16QAM	7 MHz	16 Mbps	1	4.45	5.10	4.15	3.05	10.35	11.85	4.15	7.10
			2	6.70	7.45	6.05	4.40	≫	≫	6.05	10.20
			3	8.35	9.10	7.40	5.50	≫	≫	7.40	12.75
			4	9.75	10.60	8.65	6.35	≫	≫	8.65	14.70
	14 MHz	32 Mbps	1	6.70	7.45	6.05	4.40	≫	≫	6.05	10.20
			2	9.75	10.60	8.65	6.35	≫	≫	8.65	≫

Table 17.19 Examples of 90° LMDS sector capacity limited range analysis results for some UMTS network configuration.

Environment					Sector Capacity Limited Range in km							
					Dense urban				Suburban			
UMTS loading	LMDS type	LMDS terminal type	LMDS availability %	LMDS range km	Voice 12.2 kbps	LCD 64 kbps	LCD 144 kbps	LCD 384 kbps	Voice 12.2 kbps	LCD 64 kbps	LCD 144 kbps	LCD 384 kbps
25 %		Node B range in km			0.929	0.669	0.576	0.440	2.145	1.545	1.330	1.017
	QPSK @ 26 GHz	Integrated	99.999	2.40	2.40	2.40	2.40	2.40	2.40	2.40	2.40	2.40
		antenna	99.99	3.71	3.71	3.71	3.71	***3.08***	3.71	3.71	3.71	3.71
		Antenna	99.999	3.19	3.19	3.19	3.19	***3.08***	3.19	3.19	3.19	3.19
		390 mm	99.99	5.06	5.06	5.06	***4.16***	***3.08***	5.06	5.06	5.06	5.06
		Antenna	99.999	3.99	3.99	3.99	3.99	***3.08***	3.99	3.99	3.99	3.99
		650 mm	99.99	6.48	6.48	***5.13***	***4.16***	***3.08***	6.48	6.48	6.48	6.48
	16QAM @ 26 GHz	Integrated	99.999	1.68	1.68	1.68	1.68	1.68	1.68	1.68	1.68	1.68
		antenna	99.99	2.60	2.60	2.60	2.60	2.60	2.60	2.60	2.60	2.60
		Antenna	99.999	2.23	2.23	2.23	2.23	2.23	2.23	2.23	2.23	2.23
		390 mm	99.99	3.54	3.54	3.54	3.54	3.54	3.54	3.54	3.54	3.54
		Antenna	99.999	2.79	2.79	2.79	2.79	2.79	2.79	2.79	2.79	2.79
		650 mm	99.99	4.54	4.54	4.54	4.54	***4.43***	4.54	4.54	4.54	4.54
50 %		Node B range in km			0.829	0.597	0.514	0.393	1.915	1.379	1.188	0.908
	QPSK @ 26 GHz	Integrated	99.999	2.40	2.40	2.40	2.40	2.40	2.40	2.40	2.40	2.40
		antenna	99.99	3.71	3.71	3.71	3.71	***2.75***	3.71	3.71	3.71	3.71
		Antenna	99.999	3.19	3.19	3.19	3.19	***2.75***	3.19	3.19	3.19	3.19
		390 mm	99.99	5.06	5.06	***4.58***	***3.72***	***2.75***	5.06	5.06	5.06	5.06
		Antenna	99.999	3.99	3.99	3.99	***3.72***	***2.75***	3.99	3.99	3.99	3.99
		650 mm	99.99	6.48	6.48	***4.58***	***3.72***	***2.75***	6.48	6.48	6.48	***6.37***
	16QAM @ 26 GHz	Integrated	99.999	1.68	1.68	1.68	1.68	1.68	1.68	1.68	1.68	1.68
		antenna	99.99	2.60	2.60	2.60	2.60	2.60	2.60	2.60	2.60	2.60
		Antenna	99.999	2.23	2.23	2.23	2.23	2.23	2.23	2.23	2.23	2.23
		390 mm	99.99	3.54	3.54	3.54	3.54	3.54	3.54	3.54	3.54	3.54
		Antenna	99.999	2.79	2.79	2.79	2.79	2.79	2.79	2.79	2.79	2.79
		650 mm	99.99	4.54	4.54	4.54	4.54	***3.96***	4.54	4.54	4.54	4.54

The intersection of the straight line defining the transmission density for UMTS with the curve characterising the transmission density of the LMDS sector gives the maximum sector capacity limited range (M-SCLR) for the LMDS system. This method is the most suitable for a (theoretical) deployment of UMTS base stations according to basic cellular system design methods with regular and triangular grid and hexagonal cells. For real UMTS networks, the cells with mean or maximum capacity should be considered. If only the LMDS sector transmission density is higher than the Node B transmission density, then the LMDS base station is able to serve all real UMTS base stations located within the LMDS coverage area.

The Node B transmission density has been calculated for basic UMTS propagation environments. The UMTS network design is the same as described in Section 17.6.3.1; respective calculation results are presented in Table 17.20.

The results of the Node B transmission density have been compared with some sector transmission densities for various LMDS base station configurations. The analysis was conducted for different numbers of radio channels in a single LMDS sector, QPSK and 16QAM modulation, as well as for two channel bandwidths (7 and 14 MHz). Figures 17.34 and 17.35 present these results for UMTS base stations deployed in dense urban and suburban environments, respectively, assuming a transmission density factor for a 90-degree sector and LMDS channels of 14 MHz.

It must be noted that the calculation results of the LMDS sector capacity limited range are the same for the approximation method and the system transmission density method; however, the method presented in the current section is much easier to implement, because it does not require any information about the Node B deployment and additional approximations.

17.6.3.3 Real Network Analysis Method

Both methods presented in Sections 17.6.3.1 and 17.6.3.2 allow to estimate the usefulness of various solutions that utilise radio systems with a central base station, e.g. PMP or WiMAX, for UTRAN backhaul networks; however, they presumed a UMTS network with a regular and uniform triangular grid. These methods are not suitable for real networks with irregular Node B deployment, different ranges and capacity, because they solely allow estimating the initial technical and financial effectiveness of LMDS-like systems as a transmission network. In this case, a method is needed, which allows analysing real UMTS network parameters and which find the best solution for the UTRAN transmission infrastructure. The detailed analysis of the real network has to show where an operator can expect profits using various configurations of the LMDS base stations (number of sectors, type of modulation, overlapping areas, available bandwidth and range with respect to ITU Regions, etc.). For these reasons, the comparison of implementation and operating costs of alternative solutions (i.e. PTP, PMP, LMDS, WiMAX) have to be taken into account by this method.

Typical configurations and deployments of LMDS base stations with respect to Node B locations have been depicted in Figure 17.36. In the presented cases, the number of Node Bs in neighbouring

Table 17.20 Node B transmission density in Mbps/km^2 (UMTS loading 25%).

Service	Environment		
	Dense urban	Suburban	Rural
Voice 12.2 kbps	0.8271	0.1535	0.0197
LCD 64	0.7158	0.1338	0.0173
LCD 144	1.0713	0.2009	0.0260
LCD 384	1.9636	0.3705	0.0481

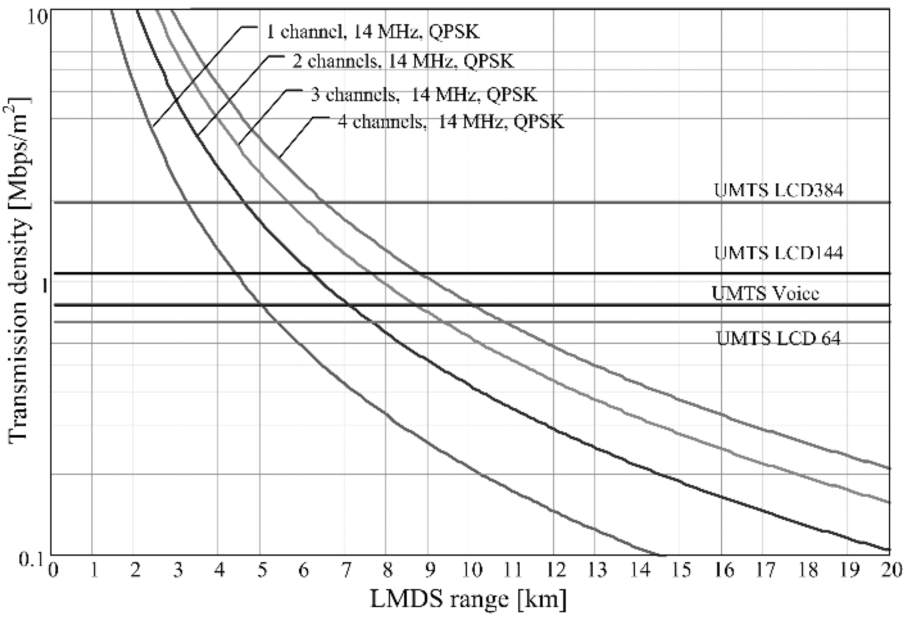


Figure 17.34 Comparison results of transmission density factor for 90° LMDS sector in dense urban area.

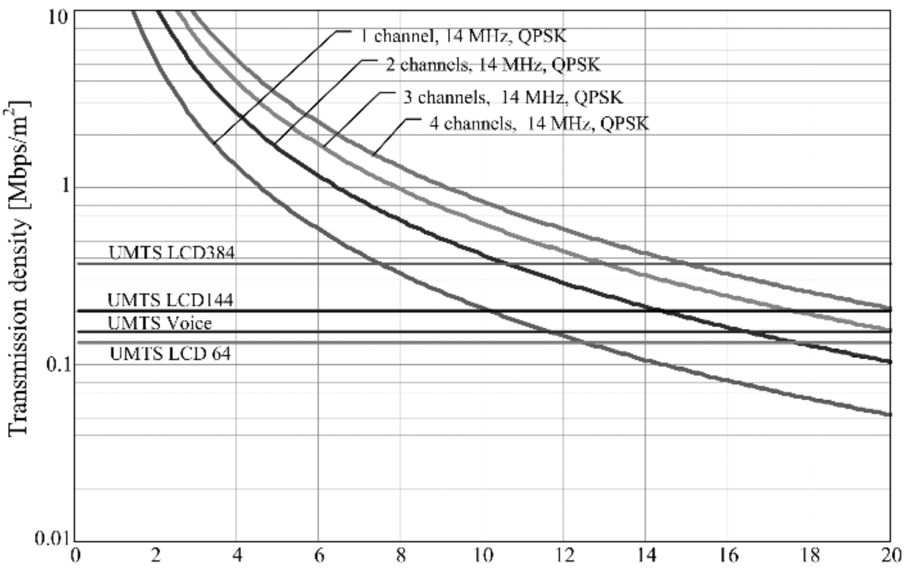


Figure 17.35 Comparison results of transmission density factor for 90° LMDS sector in suburban area.

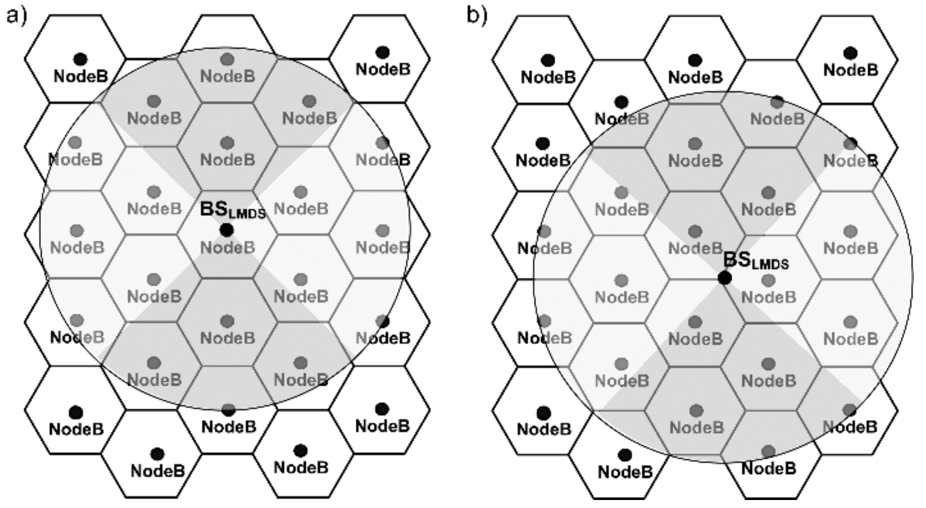


Figure 17.36 Influence of LMDS base station location on the number of served Node B: (a) collocation with Node B and (b) no collocation.

LMDS sectors can be different (from 3 up to 6 in the presented example). The sector orientations and range are of main significance, as well as the allocation procedure of Node Bs to the LMDS sectors. Thus, a generalisation of correct analysis results obtained for a single sector to the entire LMDS network can lead to incorrect results. This problem is much more complicated for real UMTS networks with non-uniform a Node B distribution.

The only method to obtain correct results is an analysis of the real LMDS base station and Node B locations, so as to get correct information about the number of Node Bs situated in each LMDS sector coverage area. Some additional parameters have to be taken into account: These concern LMDS base station configurations (sectors, modulation schemes, number of channels), coverage area and range of each LMDS sector according to the used modulation, frequency band, assumed availability and applied LMDS terminal stations. The function, which allows balancing the transmission load in adjacent LMDS sectors, should be applied as well. It should be done for Node Bs located at borders of the LMDS sector or the sector overlap area, if they are defined. It must be noted that LMDS systems vary greatly in their parameters, architecture, configuration, functionality and costs, depending on the manufacturer. Therefore, analyses have to be carried out independently for every of the manufacturer's solutions.

Some tailored computer applications are clearly necessary to study all these aspects and find some cost-effective solutions. An example of such a software [48] and some obtained cost analysis results are presented in Sections 17.6.4 and 17.6.5, respectively.

The description of other applications for automatic planning of PMP solutions for UTRAN backhaul networks as well as some calculation results for real network scenarios are presented in [49]. The authors focused on the optimisation of access networks to reduce purchase and operating costs. In the developed four-phase optimisation method, a fast heuristic algorithm has been implemented to determine the location of the PMP equipment. The algorithm uses a four-step procedure and hence solves the Weighted Independent Set (WIS) Problem for the PMP locations. Due to the underlying assumptions, the analysis was limited to Node Bs located in the same clusters and PMP base stations collocated with a Node B. PMP planning aspects or the overlapping of sectors have not been incorporated in the optimisation method.

17.6.4 COSTS EVALUATION OF UTRAN INFRASTRUCTURE – SOFTWARE EXAMPLE

For the case studies presented in the subsequent Section 17.6.5, the authors of this chapter have developed a UTRAN transmission network cost evaluation software for LMDS and PTP solutions. The software has been developed as an Excel Visual Basic for Application script, which allows configuring all required parameters of the LMDS and UMTS systems (Figure 17.37).

The software analyses the costs of real networks, hence requiring information about the real geographical Node B positions and transmission capacities. The application gives the user two possibilities to perform the calculations. The first one provides calculations for user defined data (for example coming from UMTS planning tools or real UMTS networks). The second one incorporates a simple network planning tool, which automatically distributes identical Node Bs using a regular grid. In the latter case, the UMTS base station locations depend on their range, which is defined based on the link budget for uniform services (voice transmissions at 12.2 kbps, data transmissions LCD64, LCD144, LCD384) and the Cost-Hata propagation model (only for urban areas). The user can define the service, system loading, building attenuations and the number of channels in each of three Node B's sectors. The required transmission capacity is computed as described in Section 17.2. The user can also define the percentage of the total bit rate, which is to be reserved for delivering user services. The remaining part can then be used for transmission of control information and for packet transmissions for UMTS users. If this percentage is declared as 70 %, this means that the required data bit rate, calculated from the link budget to carry out the given service by the UMTS base station, will comprise 70 % of the total bit rate. The remaining 30 % of the total bit rate can be used for other purposes, e.g. packet transmissions or control data for GSM base stations.

As a result of the above procedure, the program creates a grid of uniformly distributed UMTS base stations, forming regular hexagons with a given side length. The network size depends on the serving area of the chosen LMDS base station configuration.

In the automatic mode, the calculations can be carried out for LMDS base stations collocated with Node Bs or for the LMDS base station placed at the edge of three UMTS cells (Figure 17.36). In the

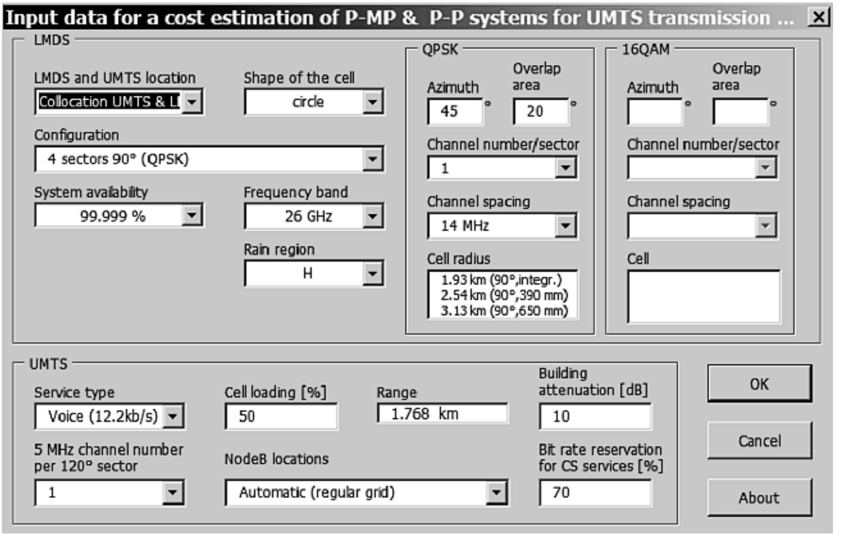


Figure 17.37 Configuration data for cost analysis.

first case, it is not necessary to use a radio data link to/from the UMTS base station, which is already at the same location as the LMDS.

The application allows also to consider the following LMDS base station configurations of a modern commercial LMDS system from one of the major manufacturers:

- base station with four sectors, all using QPSK modulation;
- base station with 12 sectors, all using QPSK modulation;
- base station with 12 sectors using QPSK modulation and four sectors using 16QAM modulation;
- base station with four sectors using QPSK modulation and four sectors using 16QAM modulation.

Choosing the configuration which uses both QPSK and 16QAM requires first to analyse the UMTS stations in the area of LMDS terminals using 16QAM (having a shorter range) and only afterwards in the area of QPSK (having a longer range). If the UMTS stations cannot be serviced by the LMDS system using 16QAM (because of the limited capacity or shorter range), then they are analysed for the possibility to be serviced by an LMDS system using QPSK.

The software application facilitates transmission loads balancing when the serviced area is divided into sectors and overlap areas are defined for adjacent sectors of LMDS. By stating appropriate angles, it is possible to independently define common areas and the angle of the first sector for each of the modulation types. Node Bs located at the sector borderline are assigned to the sector to which this borderline is to the right of the azimuth. Often, the loads of individual sectors are not equal, because of the uneven distribution of UMTS base stations within the sectors of the LMDS system. In order to overcome this, a procedure was developed allowing for sector load balancing if common areas larger then 0° are present. In the above case, stations located within predefined azimuths, excluding the common areas, are assigned to appropriate sectors. The Node Bs located within the common areas are assigned to the sector which, at the time of analysis, is less loaded. The graphical representation of the common area is shown in Figure 17.38.

The analysis can be conducted for any arbitrary number of radio channels assigned to the sectors of the LMDS system. Nevertheless, only a uniform assignment of channels is possible within a LMDS site, i.e. the same number of radio channels for each of the sectors. The selection of the number of

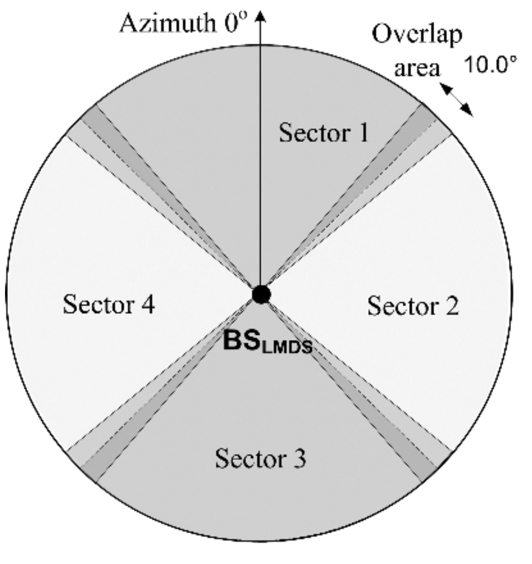


Figure 17.38 Definition of the common area and first sector azimuth.

channels is independent for systems using QPSK and 16QAM, but it must relate to the frequency band available to the operator and should rely on the guidelines for optimal use of the electromagnetic spectrum. It should be noted that, e.g., in order to have one channel in the sector of an LMDS system, the operator needs to have at least two radio channels. This rule applies when determining the extent and costs of electromagnetic spectrum utilisation.

In order for the comparison of radio resources and costs to make sense, it is also required to define the frequency requirements and the purchase costs of radio lines (PTP) required to setup links for UMTS base stations in the same configuration as defined for the LMDS system. In order to simplify the analysis, a star topology has been assumed and that each link between Node B and the UMTS RNC system is one radio link. During the analysis of the data transmission capabilities through the radio links to and from the Node B, their type is also defined for each of the links.

17.6.5 EXAMPLE CALCULATIONS AND COMPARISON OF RESULTS

The subsequent sections present some detailed case studies for cost calculations of implementing PTP and LMDS into a UTRAN backhaul network. The calculation results only take into account the real cost of equipment, excluding spectrum costs. Spectrum costs have been omitted because of differences in rentals, which depend on the country, the geographical location, service range, frequency band and assigned number of channels and their bandwidth. The calculations have been carried out for two kinds of UMTS networks. The first one is a theoretical network, which was generated by a uniform Node B deployment according to calculation results obtained for voice services. The second one is the real UMTS network, which was described and optimised in Section 15.2.

17.6.5.1 Regular Node B Deployment

All comparison results presented in this section have been obtained for UMTS network with regular and uniform triangular grid and the same size of hexagonal cells. Each Node B consists of three sectors. The Node B's requirements for the backhaul network have been calculated only for voice 12.2 kbps transmissions with a cell loading of 50 %. Additionally, it was assumed that the voice service's data rate comprises 70 % of the total bit rate. The remaining 30 % of the total bit rate are used for packet transmissions or control data. The UMTS cell size has been varied by an additional attenuation (building attenuation) in the link budget. In all cases, only LMDS base stations working in the 26 GHz frequency band and ITU rain zone E with an availability of 99.99 % have been applied.

A first analysis has been carried out for the UMTS network in non-dense urban areas, where the main problem is to have an insufficient LMDS range. In all of these three cases, the Node B's range amounts to 2.043 km. The network configurations are presented in Figure 17.39 and summarised in Table 17.21. To increase the LMDS range, the narrowest channel bandwidth has been chosen (Figure 17.39a); however, the LMDS base station capacity is poor. Thus, only less than half of all Node Bs in the LMDS range are served. These Node Bs, which are served by the LMDS base station, are drawn as hexagonals; otherwise, they are drawn as a dot. The achieved LMDS sector capacity allows to serve only two Node Bs. An increase of the LMDS capacity with additional high capacity sectors (16QAM) does not offer the required effect (case 'b'). The LMDS solution is hence more expensive than PTP radio lines, because additional equipment for LMDS base stations is needed. The limited range of the 16QAM sectors allows to serve only one or two Node Bs, and the LMDS base station capacity factor for 16QAM sectors does not exceed 30 %. The cost of the LMDS solution is a little bit lower when two out of 7 MHz radio channels are assigned to a single QPSK sector. In this case (Figure 17.39c), more LMDS equipment increases the total purchase cost.

Assigning one of the 14 MHz channels to each LMDS sector, instead of 2×7 MHz, seems to be a much better solution (Figure 17.40). It allows achieving the same LMDS base station capacity as in

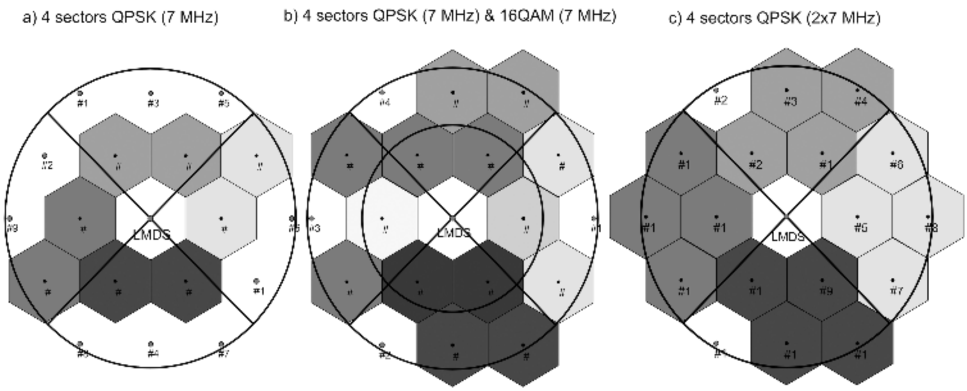


Figure 17.39 Examples of LMDS applications in UTRAN backhaul network (non-dense urban area case).

Table 17.21 Cost calculation results for LMDS and PTP applications in UTRAN backhaul network (non-dense urban area case).

Case	a		b		c	
Maximum Node B range in km	2.043		2.043		2.043	
UMTS service type	Voice (12.2 kbps)		Voice (12.2 kbps)		Voice (12.2 kbps)	
Configuration of LMDS base station	4 sect. 90° (QPSK)		4 sect. 90° (QPSK) 4 sect. 90° (16QAM)		4 sect. 90° (QPSK)	
LMDS configuration type	QPSK	16QAM	QPSK	16QAM	QPSK	16QAM
Maximum range of LMDS BS in km	7.3	—	7.3	5.10	7.300	—
Distance to the furthest serving Node B in km	6.128	—	7.076	3.538	7.076	—
Number of channels per LMDS sector	1	—	1	1	2	—
Radio channel spacing in MHz	7	—	7	7	7	—
Throughput of air interface for single radio channel in kbps	8192	—	8192	16384	8192	—
Total capacity of LMDS BS in kbps	32768	—	32768	65536	65536	—
Number of Node Bs served by LMDS	8	—	8	6	16	—
Total bit rate of served Node Bs	25894	—	25894	19421	51789	—
LMDS base station capacity factor	79.02 %	—	79.02 %	29.63 %	79.2 %	—

Table 17.21 (*continued*)

Number of not served Node Bs	10	—	4	0	2	—
Total bit rate of not served Node Bs	32368	—	12947	0	0	—
Cost of LMDS Base station	**€72 853**		**€160 471**		**€138 423**	
Cost of LMDS terminals	**€28 752**	—	**€28 752**	**€30 222**	**€43 128**	—
Total cost of LMDS equipment	**€101 605**		**€219 445**		**€195 927**	
Total cost of PTP equipment	**€99 217**		**€173 629**		**€198 433**	

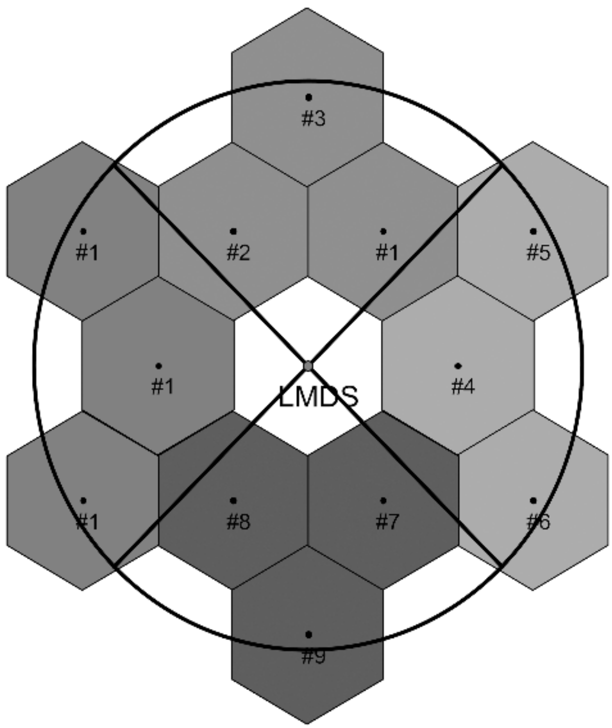


Figure 17.40 Cost-optimised LMDS solution for UTRAN backhaul network in non-dense area.

presented case 'c' with only a slightly decreased range. This solution does not require additional base station modules (compared to case 'a') and hence limits the LMDS system cost to 115 981€. This is competitive to the 148 825€ for the respective PTP radio line costs. A higher capacity allows serving 12 Node Bs; however, because of the limited range, the LMDS base station capacity factor does not exceed 60 %. For this reason, it is recommended to use higher range LMDS systems working in the 3.5 and 10 GHz bands in the case of suburban and rural areas.

Above analysis shows that if there is no more than two Node Bs served with one radio channel in each LMDS sector, the LMDS based solution is not competitive to the PTP one (for an assumed LMDS equipment vendor). On the other hand, the LMDS base station capacity factor is low; thus, if remaining LMDS resources can be used for other purposes, the cost-effectiveness of this solution can be increased.

Let us also consider an LMDS system used in a UMTS dense urban environment. With reference to the results presented in Section 17.6.3, the cases with 25 % of UMTS loading and a Node B range of 929 m assuming voice services will be analysed here. The results are presented in Table 17.22 and Figure 17.41.

Usage of the 14 MHz channel does not assure the required LMDS system capacity, so as to serve all Node Bs within its range (the same situation was for non-dense urban areas). However, this solution

Table 17.22 Cost calculation results for LMDS and PTP applications in UTRAN backhaul network (dense urban area case).

Case	a		b		c	
Maximum Node B range in km	0.929		0.929		0.929	
UMTS service type	Voice (12.2 kbps)		Voice (12.2 kbps)		Voice (12.2 kbps)	
Configuration of LMDS base station	4 sect. 90° (QPSK)		4 sect. 90° (QPSK) 4 sect. 90° (16QAM)		4 sect. 90° (QPSK)	
LMDS configuration type	QPSK	16QAM	QPSK	16QAM	QPSK	16QAM
Maximum range of LMDS BS in km	6.480	—	6.480	5.110	6.480	—
Distance to the furthest serving Node B in km	5.571	—	6.433	4.825	6.433	—
Number of channels per LMDS sector	1	—	1	1	2	—
Radio channel spacing in MHz	14	—	14	7	14	—
Throughput of air interface for single radio channel in kbps	16384	—	16384	16384	16384	—
Total capacity of LMDS BS in kbps	65536	—	65536	65536	131072	—
Number of Node Bs served by LMDS	40	—	24	36	60	—
Total bit rate of served Node Bs	64736	—	38841	58262	97104	—
LMDS base station capacity factor in %	98.78	—	59.27	88.90	74.08	—
Number of not served Node Bs	20	—	0	0	0	—
Total bit rate of not served Node Bs	32368	—	0	0	0	—

Table 17.22 (*continued*)

Cost of LMDS Base station	**€72 853**		**€160 471**		**€138 423**	
Cost of LMDS terminals	**€143 759**	—	**€86 255**	**€181 334**	**€215 638**	—
Total cost of LMDS equipment	**€216 612**		**€428 060**		**€354 061**	
Total cost of PTP equipment	**€496 083**		**€744 124**		**€744 124**	

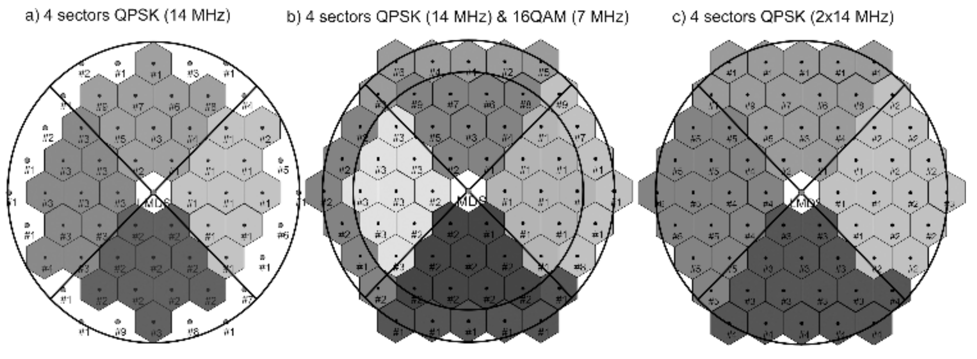


Figure 17.41 Examples of LMDS and PTP applications in UTRAN backhaul network (non-dense urban area case).

is about twice as cheap as PTP radio lines used in UTRAN. Usage of two channels per LMDS sector seems to be reasonable in the case of a uniform Node B distribution, because of the significantly lower costs of equipment purchase. Usage of 16QAM guarantees sufficient transmission reserves for potential high speed services or for utilisation by other systems belonging to the same operator (e.g. GSM, WiFi access points, etc.).

It should be noted that in analysed case (case 'b' in Table 17.22), 7 MHz channels for 16QAM have been used. As for QPSK, at least 14 MHz channels should be used in dense urban areas; transmission capacity otherwise deteriorates.

17.6.5.2 Real Network Case

To present the real case scenario, an analysis for a network based on the case study from Section 15.2 was done. Figure 17.42 presents a UMTS network layout with LMDS sectors on top of it (QPSK and 16QAM sectors). The LMDS base stations were collocated with Node Bs. The results are presented in Table 17.23. The total cost of equipment for LMDS compared to PTP microwave radio lines is about 30 % lower, which is a significant gain.

In summary, use of Point-to-MultiPoint systems can give significant savings in the costs of UTRAN transmission infrastructures, particularly in urban areas. As every wireless system, also PMP must be planned carefully to achieve the best results and a high spectrum utilisation. Even greater savings can be expected when WiMAX will overcome synchronisation issues (see Section 17.5.2) and will become mature enough to be used in the same way as LMDS systems. Finally, it is clearly an interesting idea to have one wireless system (UMTS) served by another wireless system (PMP), where UMTS base

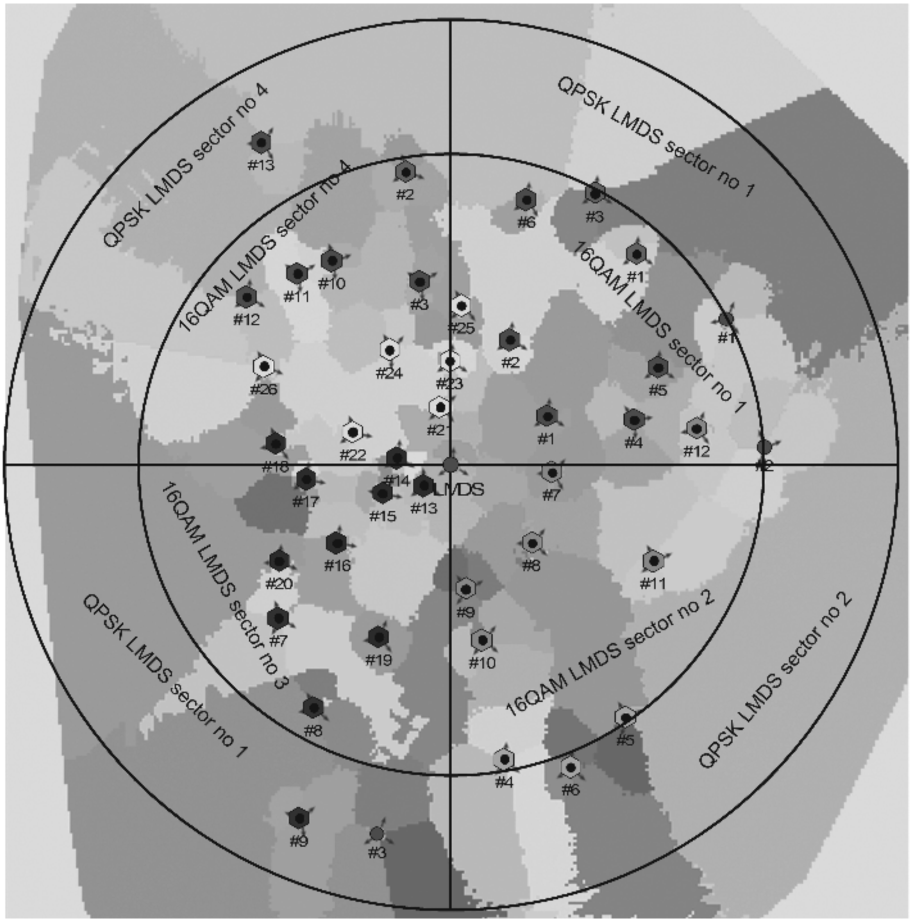


Figure 17.42 LMDS applications in real UTRAN backhaul network (collocation with Node B).

Table 17.23 Cost calculation results for LMDS and PTP applications in UTRAN backhaul network (real network case).

Configuration of LMDS base station	4 sectors 90° (QPSK) and 4 sect. 90° (16QAM)	
LMDS sector configuration	QPSK	16QAM
Maximum range of LMDS BS in km	3.99	2.79
Distance to the furthest serving Node B in km	3.446	2.474
Number of channels per LMDS sector	1	1
Radio channel spacing in MHz	14	14

Table 17.23 (*continued*)

Throughput of air interface for single radio channel in kbps	16384	32768
Total capacity of LMDS BS in kbps	65536	131072
Number of Node Bs served by LMDS	13	26
Total bit rate of served Node Bs	58054	122016
LMDS base station capacity factor in %	88.58	93.09
Number of not served Node Bs	3	0
Total bit rate of not served Node Bs	21983	51470
Cost of LMDS Base station	€160 471	
Cost of LMDS terminals	€46 722	€130 964
Total cost of LMDS equipment	**€338 156**	
Total cost of PTP equipment	**€490 704**	

stations become terminals for PMP. Let this idea guide the operators' transmission departments to the development of a more efficient and cheaper network infrastructure.

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