SKP Engineering College

Tiruvannamalai – 606611

A Course Material

on

Wireless Networks

By

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Electronics and Communication Engineering Department

Quality Certificate

This is to Certify that the Electronic Study Material

Subject Code: EC6802

Subject Name:Wireless Networks

Year/Sem:IV/VIII

Being prepared by me and it meets the knowledge requirement of the University curriculum.

Signature of the Author

Name: S.Baskaran

Designation: Professor

This is to certify that the course material being prepared by Dr.S.Baskaran is of the adequate quality. He has referred more than five books and one among them is from abroad author.

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Signature of the Principal

Name: Dr.V.Subramania Bharathi

Seal:

EC6802 WIRELESS NETWORKS L T P C 3 0 0 3

OBJECTIVES:

To study about Wireless networks, protocol stack and standards.

To study about fundamentals of 3G Services, its protocols and applications.

To study about evolution of 4G Networks, its architecture and applications.

UNIT I WIRELESS LAN 9

Introduction-WLAN technologies: Infrared, UHF narrowband, spread spectrum - IEEE802.11: System architecture, protocol architecture, physical layer, MAC layer, 802.11b, 802.11a – Hiper LAN: WATM, BRAN, HiperLAN2 – Bluetooth: Architecture, Radio Layer, Baseband layer, Link manager Protocol, security – IEEE802.16-WIMAX: Physical layer, MAC, Spectrum allocation for WIMAX

UNIT II MOBILE NETWORK LAYER 9

Introduction – Mobile IP: IP packet delivery, Agent discovery, tunneling and encapsulation, IPV6-Network layer in the internet- Mobile IP session initiation protocol – mobile ad-hoc network: Routing, Destination Sequence distance vector, Dynamic source routing

UNIT III MOBILE TRANSPORT LAYER 9

TCP enhancements for wireless protocols – Traditional TCP: Congestion control, fast retransmit/fast recovery, Implications of mobility – Classical TCP improvements: Indirect TCP, Snooping TCP, Mobile TCP, Time out freezing, Selective retransmission, Transaction oriented TCP – TCP over 3G wireless networks**.**

UNIT IV WIRELESS WIDE AREA NETWORK 9

Overview of UTMS Terrestrial Radio access network-UMTS Core network Architecture: 3G-MSC, 3G-SGSN, 3G-GGSN, SMS-GMSC/SMS-IWMSC, Firewall, DNS/DHCP-High speed Downlink packet access (HSDPA)- LTE network architecture and protocol.

UNIT V 4G NETWORKS 9

Introduction – 4G vision – 4G features and challenges – Applications of 4G – 4G Technologies: Multicarrier Modulation, Smart antenna techniques, OFDM-MIMO systems, Adaptive Modulation and coding with time slot scheduler, Cognitive Radio.

TOTAL: 45 PERIODS

OUTCOMES: Upon completion of the course, the students will be able to

Conversant with the latest 3G/4G and WiMAX networks and its architecture.

Design and implement wireless network environment for any application using latest wireless protocols and standards.

Implement different type of applications for smart phones and mobile devices with latest network strategies.

TEXT BOOKS:

1.Jochen Schiller, ”Mobile Communications”, Second Edition, Pearson Education 2012. (Unit I,II,III)

2.Vijay Garg , “Wireless Communications and networking”, First Edition, Elsevier 2007. (Unit IV,V)

REFERENCES:

1. Erik Dahlman, Stefan Parkvall, Johan Skold and Per Beming, “3G Evolution HSPA and LTE for Mobile Broadband”, Second Edition, Academic Press, 2008.

2. Anurag Kumar, D.Manjunath, Joy kuri, “Wireless Networking”, First Edition, Elsevier 2011.

3. Simon Haykin , Michael Moher, David Koilpillai, “Modern Wireless Communications”

CONTENTS

Unit - I

Wireless LAN

Part - A

1. What are the advantages of wireless LAN? (L-1,CO-1)

1.Flexibility

2. Planning

3.Robustness

4.Design

2. What are the properties of ISM band?(L-2,CO-1)

Frequency of operation 902-928 MHZ 2.4-2.483 GHZ 5.725-5.875 GHZ

Transmit power limitation of 1 watt for DSSS and FHSS low power with any modulation.

3. **Mention the three basic rules (or) etiquette of spectrum. (H-2,CO-1)**

1.Listen Before talk

2. Low Transmit Power

3. Restricted duration fr transmission

4. **State the features of wireless LAN. (H-3,CO-1)**

1. Power management to save the battery power.

2. The handling of hidden nodes.

3. The ability to operate world wide**.**

5. Draw the frame format of IEEE 802.11 physical layer using FHSS. (L-1,CO-1)

Sync	SFD	PLW	PSF	HEC	MPDU
(80)	(16)	(12)	(4)	(16)	variable

6. **List the type of architecture used in IEEE 802.11. (H-3,CO-1)**

1. Infrastructure Based

2. Adhoc based

7. **What are the characteristics of DSSS? (H-1,CO-1)**

1. Robustness against interface
2. Insensitivity to multipath propoagation
3. Implementation is complex compared to FHSS

8. What is the formula used in DSSS and FHSS to scramble the transmitted bits? (L-2,CO-1)

$S(Z)=Z^7+Z^4+1$ for dc blocking and whitening of spectrum.

9. What is meant by wireless ATML? (L-1,CO-1)

Wireless ATM is sometimes called as mobile ATM or WATM. It does not only describe a transmission technology, but specify a complete communication system. It develops a set of specifications that extends the use of ATM technology to wireless network.

10. **What are the possibilities of communication between mobile terminal and a fixed terminal?**

(L-1,CO-1)

1. WLAN to LAN

2. WLAN to ATM

3.WATM to ATM

11. **What are the versions of HIPER LAN? (L-1,CO-1)**

1. HIPER LAN 1

2. HIPER LAN 2

3.HIPER Access

4. HIPER Link

12. **List the protocols used in HIPER LAN-2. (H-2,CO-1)**

1.Radio Link Protocol.
2. DLC connection Protocol
3. Radio Resource Protocol
4. Association control Function**.**

13. **What is meant by data link layer? (H-3,CO-1)**

Data Link Layer(DLC) provides the logical link between an access point and the mobile terminals over the OFDM physical layer.

14. **What are the differences between the 802.11a and HIPET LAN-2? (L-3,CO-1)**

The HIPER LAN-2 standard uses the same physical layer as 802.11a with a MAC that supports the needs of the cellular telephone industry is supporting mechanisms for tariff, integration with existing cellular systems and providing QOS. IEEE 802.11camp is a connectionless WLAN camp tat evolved from data oriented computer communications. HIPER LAN-2 camp is connection based WLANs addressing the needs of voice oriented cellular telephone.

15. **State the relationship between HYPER LAN-2 and WATM. (H-2,CO-1)**

HIPER LAN-2 aims at higher data rates and intends to accommodate ATM as well as IP type access.

16. **What do you mean by WPA? (L-1,CO-1)**

The 802.11itasks group has developed a set of capabilities to address the WLAN security capabilities to address the WLAN security issues. In order to accelerate the introduction of strong security into WLANs, the WI-FI alliance promulgated WI-FI Protected Access(WPA) as WI-FI standard. WPA is a set of security mechanisms that eliminate most 802.11 srcurity issues and was based on the current state of th 802.11i standard.

PART B

1.Brief explain about the advantages of WLAN Techniques. (L-1,CO-1)

● Flexibility**:** Within radio coverage, nodes can communicate without further restriction. Radio waves can penetrate walls, senders and receivers can be placed anywhere (also non-visible, e.g., within devices, in walls etc.). Sometimes wiring is difficult if firewalls

separate buildings (real firewalls made out of, e.g., bricks, not routers set up as a firewall). Penetration of a firewall is
only permitted at certain points to prevent fire from spreading too fast.

● **Planning:** Only wireless ad-hoc networks allow for communication without previous planning, any wired network needs wiring plans. As long as devices follow the same standard, they can communicate. For wired networks, additional cabling with the right plugs and probably interworking units (such as switches) have to be provided.

● **Design:** Wireless networks allow for the design of small, independent devices which can for example be put into a pocket. Cables not only restrict users but also designers of small PDAs, notepads etc. Wireless senders and receivers can be hidden in historic buildings, i.e., current networking technology can be introduced without being visible.

● **Robustness:** Wireless networks can survive disasters, e.g., earthquakes or users pulling a plug. If the wireless devices survive, people can still communicate. Networks requiring a wired infrastructure will usually break down.

● **Cost:** After providing wireless access to the infrastructure via an access point for the first user, adding additional users to a wireless network will not increase the cost. This is, important for e.g., lecture halls, hotel lobbies or gate areas in airports where the numbers using the network may vary significantly. Using a fixed network, each seat in a lecture hall should have a plug
for the network although many of them might not be used permanently. Constant plugging and unplugging will sooner or later destroy the plugs. Wireless connections do not wear out. But WLANs also have several **disadvantages**:

● **Quality of service:** WLANs typically offer lower quality than their wired counterparts. The main reasons for this are the lower bandwidth due to limitations in radio transmission (e.g., only 1–10 Mbit/s user data rate instead of 100–1,000 Mbit/s), higher

error rates due to interference (e.g., 10–4 instead of 10–12 for fiber optics), and higher delay/delay variation due to extensive error correction and detection mechanisms.

● **Proprietary solutions:** Due to slow standardization procedures, many companies have come up with proprietary solutions offering standardized functionality plus many enhanced features (typically a higher bit rate using a patented coding technology or special inter-access point protocols). However, these additional features only work in a homogeneous environment, i.e., when adapters from the same vendors are used for all wireless nodes. At least most components today adhere to the basic standards IEEE 802.11b or (newer) 802.11a (see section 7.3).

● **Restrictions:** All wireless products have to comply with national regulations. Several government and non-government institutions worldwide regulate the operation and restrict frequencies to minimize interference. Consequently, it takes a very long time to establish global solutions like, e.g., IMT-2000, which comprises many individual standards. WLANs are limited to low-power senders and certain license-free frequency bands, which are not the same worldwide.

● **Safety and security:** Using radio waves for data transmission might interfere with other high-tech equipment in, e.g., hospitals. Senders and receivers are operated by laymen and, radiation has to be low. Special precautions have to be taken to prevent safety hazards. The open radio interface makes eavesdropping much easier in WLANs than, e.g., in the case of fiber optics. All standards must offer (automatic) encryption, privacy mechanisms, support for anonymity etc. Otherwise more and more wireless networks will be hacked into as is the case already (aka war driving: driving around looking for unsecured wireless networks; WarDriving, 2002).

Many different, and sometimes competing, design goals have to be taken into account for LANs to ensure their commercial success:

● **Global operation:** WLAN products should sell in all countries so, national and international frequency regulations have to be considered. In contrast to the infrastructure of wireless WANs, LAN equipment may be carried from one country into another – the operation should still be legal in this case.

● **Low power:** Devices communicating via a WLAN are typically also wireless devices running on battery power. The LAN design should take this into account and implement special power-saving modes and power management functions. Wireless communication with devices plugged into a power outlet is only useful in some cases (e.g., no additional cabling should be necessary
for the network in historic buildings or at trade shows). However, the future clearly lies in small handheld devices without any restricting wire.

● **License-free operation:** LAN operators do not want to apply for a special license to be able to use the product. The equipment must operate in a license-free band, such as the 2.4 GHz ISM band.

● **Robust transmission technology:** Compared to their wired counterparts, WLANs operate under difficult conditions. If they use radio transmission, many other electrical devices can interfere with them (vacuum cleaners, hairdryers, train engines etc.). WLAN transceivers cannot be adjusted for perfect transmission in a standard office or production environment. Antennas
are typically omnidirectional, not directed. Senders and receivers may move.

● **Simplified spontaneous cooperation:** To be useful in practice, WLANs should not require complicated setup routines but should operate spontaneously after power-up. These LANs would not be useful for supporting, e.g., ad-hoc meetings.
● **Easy to use:** In contrast to huge and complex wireless WANs, wireless LANs are made for simple use. They should not require complex management, but rather work on a plug-and-play basis.

● **Protection of investment:** A lot of money has already been invested into wired LANs. The new WLANs should protect this investment by being interoperable with the existing networks. This means that simple bridging between the different LANs should be enough to interoperate, i.e., the wireless LANs should support the same data types and services that standard
LANs support.

● **Safety and security:** Wireless LANs should be safe to operate, especially regarding low radiation if used, e.g., in hospitals. Users cannot keep safety distances to antennas. The equipment has to be safe for pacemakers, too. Users should not be able to read personal data during transmission, i.e., encryption mechanisms should be integrated. The networks should also
take into account user privacy, i.e., it should not be possible to collect roaming profiles for tracking persons if they do not agree. Wireless LAN 203

● **Transparency for applications:** Existing applications should continue to run over WLANs, the only difference being higher delay and lower bandwidth. The fact of wireless access and mobility should be hidden if it is not relevant, but the network should also support location aware applications, e.g., by providing location information. The following sections first introduce basic transmission technologies used for WLANs, infra red and radio, then the two basic settings for WLANs: infrastructure- based and ad-hoc, are presented. The three main sections of this
present the IEEE standard for WLANs, IEEE 802.11, the European ETSI standard for a high-speed WLAN with QoS support, HiperLAN2, and finally, an industry approach toward wireless personal area networks (WPAN), i.e., WLANs at an even smaller range, called Bluetooth.

2. Compare Infra red vs radio transmission techniques. (L-2,CO-1)

Today, two different basic transmission technologies can be used to set up WLANs. One technology is based on the transmission of infra red light (e.g., at 900 nm

wavelength), the other one, which is much more popular, uses radio transmission in the GHz range (e.g., 2.4 GHz in the license-free ISM band). Both technologies can be used to set up ad-hoc connections for work groups, to connect, e.g., a desktop with a printer without a wire, or to support mobility within a small area.

Infra red technology uses diffuse light reflected at walls, furniture etc. or directed light if a line-of-sight (LOS) exists between sender and receiver. Senders can be simple light emitting diodes (LEDs) or laser diodes. Photodiodes act as receivers. Details about infra red technology, such as modulation, channel impairments etc. can be found in Wesel (1998) and Santamaría (1994).

● The main **advantages** of infra red technology are its simple and extremely cheap senders and receivers which are integrated into nearly all mobile devices available today. PDAs, laptops, notebooks, mobile phones etc. have an infra red data association (IrDA) interface. Version 1.0 of this industry standard implements data rates of up to 115 kbit/s, while IrDA 1.1 defines
higher data rates of 1.152 and 4 Mbit/s. No licenses are needed for infra red technology and shielding is very simple. Electrical devices do not interfere with infra red transmission.

● **Disadvantages** of infra red transmission are its low bandwidth compared to other LAN technologies. Typically, IrDA devices are internally connected to a serial port limiting transfer rates to 115 kbit/s. Even 4 Mbit/s is not a particularly high data rate. However, their main disadvantage is that infra red is quite easily shielded. Infra red transmission cannot penetrate walls or other obstacles. Typically, for good transmission quality and high data rates
a LOS, i.e., direct connection, is needed.

Almost all networks described in this book use **radio** waves for data transmission, e.g., GSM at 900, 1,800, and 1,900 MHz, DECT at 1,880 MHz etc.

● **Advantages** of radio transmission include the long-term experiences made with radio transmission for wide area networks (e.g., microwave links) and mobile cellular phones. Radio transmission can cover larger areas and can penetrate (thinner) walls, furniture, plants etc. Additional coverage is gained by reflection. Radio typically does not need a LOS if the frequencies are not too high. Furthermore, current radio-based products offer much higher transmission rates (e.g., 54 Mbit/s) than infra red (directed laser links, which offer data rates well above 100 Mbit/s. These are not considered here as it is very difficult to use them with mobile devices).

● Again, the main advantage is also a big **disadvantage** of radio transmission. Shielding is not so simple. Radio transmission can interfere with other senders, or electrical devices can destroy data transmitted via radio. Additionally, radio transmission is only permitted in certain frequency

bands. Very limited ranges of license-free bands are available worldwide and those that are available are not the same in all countries. However, a lot of harmonization is going on due to market pressure. Of the three WLAN technologies presented, only one (IEEE 802.11) standardized infra red transmission in addition to radio transmission. The other two (HIPERLAN and Bluetooth) rely on radio. The main reason for this are the shielding problems of infra red. WLANs should, e.g., cover a whole floor of a building and not just the one room where LOSs exist. Future mobile devices may have to communicate while still in a pocket or a suitcase so cannot rely on infra red. The big advantage of radio transmission in everyday use is indeed the ability to penetrate certain materials and that a LOS is not required. Many users experience a lot of difficulties adjusting infra red ports of, e.g., mobile phones to the infra red port of their PDA. Using, e.g., Bluetooth is much simpler.

3. Explain about Infrastructure and ad-hoc networks in detail. (L-1,CO-1)

Many WLANs of today need an **infrastructure** network. Infrastructure networks not only provide access to other networks, but also include forwarding functions, medium access control etc. In these infrastructure-based wireless networks, communication typically takes place only between the wireless nodes and the access point (see Figure 7.1), but

not directly between the wireless nodes. The access point does not just control medium access, but also acts as a bridge to other wireless or wired networks. Figure 7.1 shows three access points with their three wireless networks and a wired network. Several wireless networks may form one logical wireless network, so the access points together with the fixed network in between can connect several wireless networks to form a larger network beyond actual radio coverage.

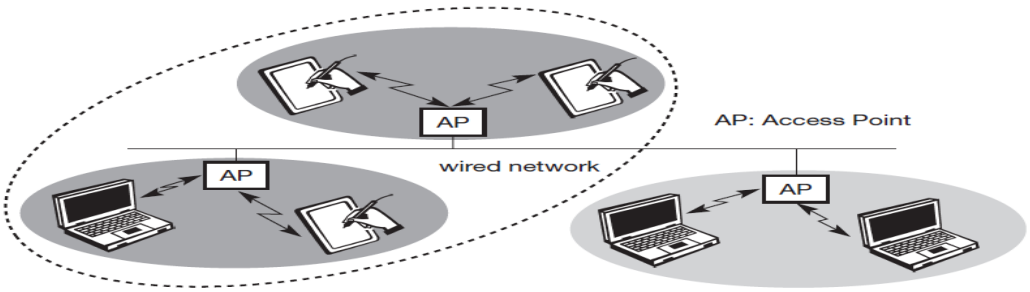


Figure 7.1 Example of three infrastructure-based wireless networks

Typically, the design of infrastructure-based wireless networks is simpler because most of the network functionality lies within the access point, whereas the wireless clients can remain quite simple. This structure is reminiscent of switched Ethernet or other star-based networks, where a central element (e.g., a switch) controls network flow. This type of network can use different access schemes with or without collision. Collisions may occur if medium access of the wireless nodes and the access point is not coordinated. However, if only the access point controls medium access, no collisions are possible. This setting may be useful for quality of service guarantees such as minimum bandwidth for certain nodes. The access point may poll the single wireless nodes to ensure the data rate. Infrastructure-based networks lose some of the flexibility wireless networks can offer, e.g., they cannot be used for disaster relief in cases where no infrastructure

is left. Typical cellular phone networks are infrastructure-based networks for a wide area. Also satellite-based cellular phones have an infrastructure – the satellites . Infrastructure does not necessarily imply a wired fixed network.

Ad-hoc wireless networks, however, do not need any infrastructure to work. Each node can communicate directly with other nodes, so no access point controlling medium

access is necessary. Figure 7.2 shows two ad-hoc networks with three nodes each. Nodes within an ad-hoc network can only communicate if they can reach each other physically, i.e., if they are within each other's radio range or if other nodes can forward the message. Nodes from the two networks shown in Figure 7.2 cannot, therefore, communicate with each other if they are not within the same radio range. In ad-hoc networks, the complexity of each node is higher because every node has to implement medium access mechanisms, mechanisms to handle hidden or exposed terminal problems, and perhaps priority mechanisms, to provide a certain quality of service. This type of wireless network exhibits the greatest possible flexibility as it is, for example, needed for unexpected meetings, quick replacements of infrastructure or communication scenarios far away from any infrastructure.

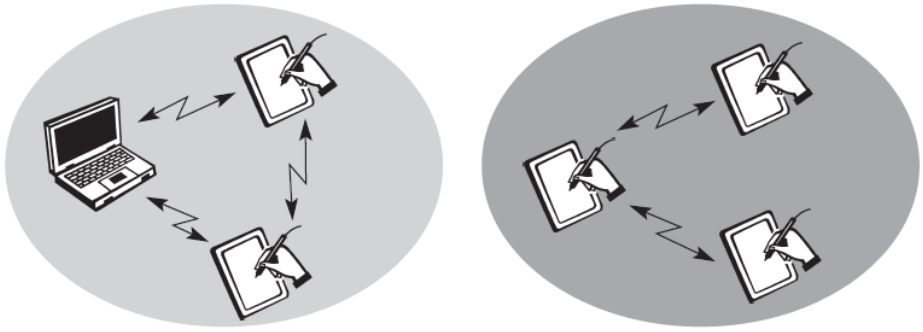


Figure 7.2
Example of two ad-hoc wireless networks

Clearly, the two basic variants of wireless networks (here especially WLANs), infrastructure-based and ad-hoc, do not always come in their pure form. There are networks that rely on access points and infrastructure for basic services (e.g., authentication of access, control of medium access for data with associated quality of service, management functions), but that also allow for direct communication between the wireless nodes.

However, ad-hoc networks might only have selected nodes with the capabilities of forwarding data. Most of the nodes have to connect to such a special node first to transmit data if the receiver is out of their range. From the three WLANs presented, IEEE 802.11 (see section 7.3) and HiperLAN2 (see section 7.4) are typically infrastructure-based networks, which additionally support ad-hoc networking. However,

many implementations only offer the basic infrastructure-based version. The third WLAN, Bluetooth (see section 7.5), is a typical wireless ad-hoc network. Bluetooth focuses precisely on spontaneous ad-hoc meetings or on the simple connection of two or more devices without requiring the setup of an infrastructure.

4. Explain in detail about IEEE 802.11 (L-3,CO-1)

The IEEE standard 802.11 (IEEE, 1999) specifies the most famous family of WLANs in which many products are available. As the standard's number indicates, this standard belongs to the group of 802.x LAN standards, e.g., 802.3 Ethernet or 802.5 Token Ring. This means that the standard specifies the physical and medium access layer adapted to the special requirements of wireless LANs, but offers the same interface as the others to higher layers to maintain interoperability. The primary goal of the standard was the specification of a simple and robust

WLAN which offers time-bounded and asynchronous services. The MAC layer should be able to operate with multiple physical layers, each of which exhibits a different medium sense and transmission characteristic. Candidates for physical layers were infra red and spread spectrum radio transmission techniques.

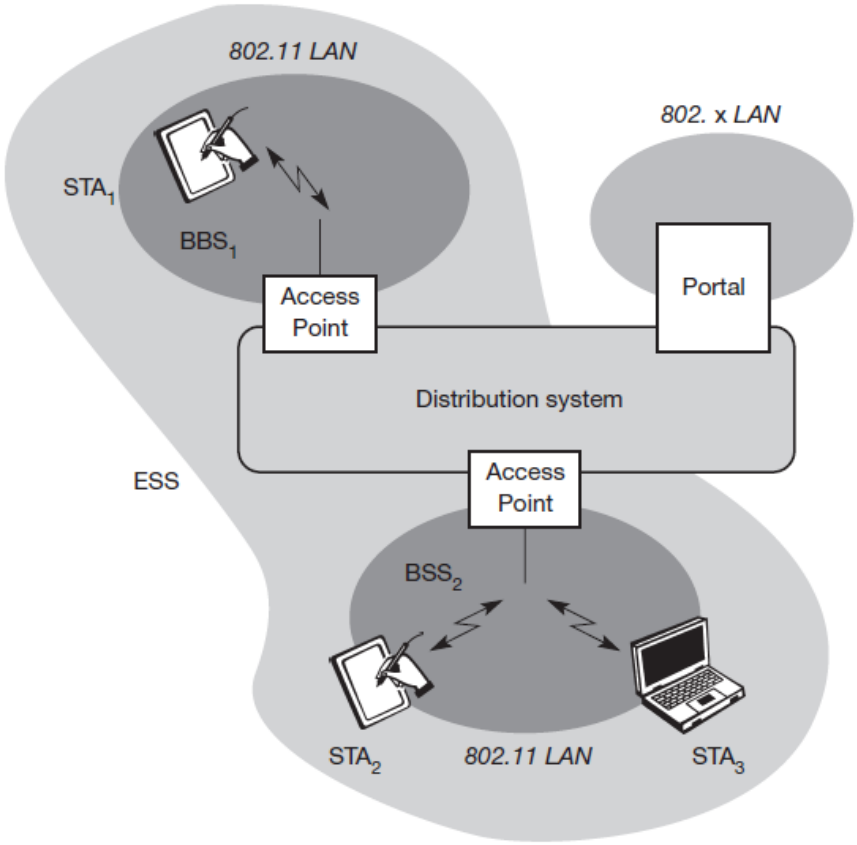
Additional features of the WLAN should include the support of power management to save battery power, the handling of hidden nodes, and the ability to operate worldwide. The 2.4 GHz ISM band, which is available in most countries around the world, was chosen for the original standard. Data rates envisaged for the standard were 1 Mbit/s mandatory and 2 Mbit/s optional.

The following sections will introduce the system and protocol architecture of the initial IEEE 802.11 and then discuss each layer, i.e., physical layer and medium access. After that, the complex and very important management functions of the standard are presented. Finally, this subsection presents the enhancements of the original standard for higher data rates, 802.11a (up to 54 Mbit/s at 5 GHz) and 802.11b (today the most successful with 11 Mbit/s) together with further developments for security support, harmonization, or other modulation schemes.

7.3.1 System architecture Wireless networks can exhibit two different basic system architectures as shown in section 7.2: infrastructure-based or ad-hoc. Figure 7.3 shows

the components of an infrastructure and a wireless part as specified for IEEE 802.11. Several nodes, called **stations (STAi)**, are connected to **access points (AP)**. Stations are terminals with access mechanisms to the wireless medium and radio contact to

Figure 7.3 Architecture of an infrastructure-based IEEE 802.11



the AP. The stations and the AP which are within the same radio coverage form a **basic service set (BSSi)**. The example shows two BSSs – BSS1 and BSS2 – which are connected via a **distribution system**. A distribution system connects several BSSs via the AP to form a single network and thereby extends the wireless coverage area. This network is now called an **extended service set (ESS)** and has its own identifier, the ESSID. The ESSID is the 'name' of a network and is used to separate different networks. Without knowing the ESSID (and assuming no

hacking) it should not be possible to participate in the WLAN. The distribution system connects the wireless networks via the APs with a **portal**, which forms the interworking unit to other LANs. The architecture of the distribution system is not specified further in IEEE 802.11. It could consist of bridged IEEE LANs, wireless links, or any other

networks. However, **distribution system services** are defined in the standard (although, many products today cannot interoperate and needs the additional standard IEEE 802.11f to specify an inter access point protocol, see section 7.3.8). Stations can select an AP and associate with it. The APs support roaming (i.e., changing access points), the distribution system handles data transfer between the different APs. APs provide synchronization within a BSS, support power management, and can control medium access to support time-bounded service. These and further functions are explained in the following sections. In addition to infrastructure-based networks, IEEE 802.11 allows the building of ad-hoc networks between stations, thus forming one or more independent

BSSs (IBSS) as shown in Figure 7.4. In this case, an IBSS comprises a group of stations using the same radio frequency. Stations STA1, STA2, and STA3 are in IBSS1, STA4 and STA5 in IBSS2. This means for example that STA3 can communicate

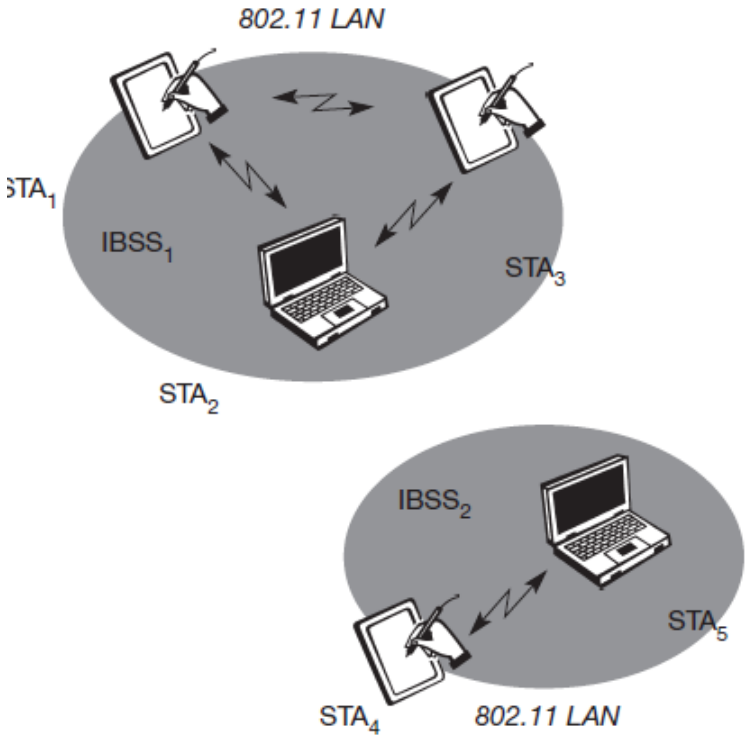


Figure 7.4 Architecture of IEEE 802.11 ad-hoc wireless LANs

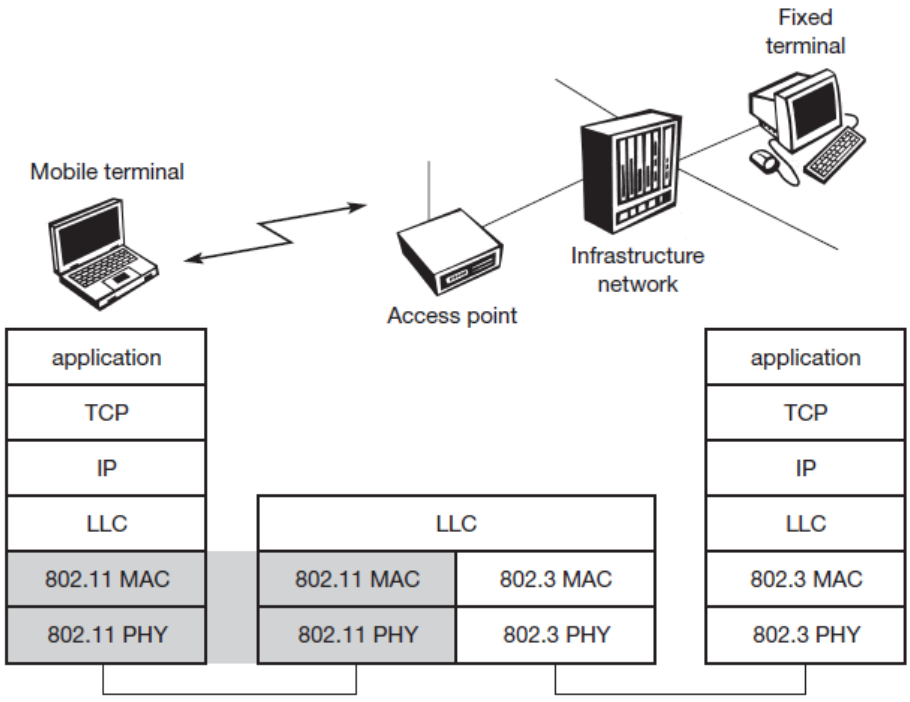
directly with STA2 but not with STA5. Several IBSSs can either be formed via the distance between the IBSSs (see Figure 7.4) or by using different carrier frequencies (then the IBSSs could overlap physically). IEEE 802.11 does not specify any special

nodes that support routing, forwarding of data or exchange of topology information as, e.g., HIPERLAN 1 (see section 7.4) or Bluetooth (see section 7.5).

Protocol architecture

As indicated by the standard number, IEEE 802.11 fits seamlessly into the other 802.x standards for wired LANs (see Halsall, 1996; IEEE, 1990). Figure 7.5 shows the most common scenario: an IEEE 802.11 wireless LAN connected to a switched IEEE 802.3 Ethernet via a bridge. Applications should not notice any difference apart from the lower bandwidth and perhaps higher access time from the wireless LAN. The WLAN behaves like a slow wired LAN. Consequently, the higher layers (application, TCP, IP) look the same for wireless nodes as for wired nodes. The upper part of the data link control layer, the logical link control (LLC), covers the differences of the medium access control layers needed for the different media. In many of today's networks, no explicit LLC layer is visible. Further details like Ethertype or sub-network access protocol (SNAP) and bridging technology are explained in, e.g., Perlman (1992). The IEEE 802.11 standard only covers the physical layer **PHY** and medium access layer **MAC** like the other 802.x LANs do. The physical layer is subdivided into the **physical layer convergence protocol (PLCP)** and the **physical medium dependent** sublayer **PMD** (see Figure 7.6). The basic tasks of the MAC layer comprise medium access, fragmentation of user data, and encryption. The

Figure 7.5
IEEE 802.11 protocol architecture and bridging



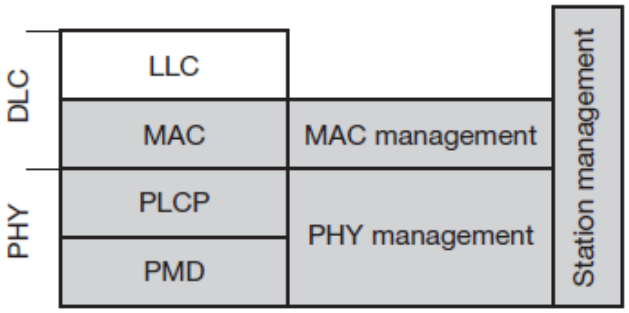


Figure 7.6
Detailed IEEE 802.11 protocol architecture and management

PLCP sublayer provides a carrier sense signal, called clear channel assessment (CCA), and provides a common PHY service access point (SAP) independent of the transmission technology. Finally, the PMD sublayer handles modulation and encoding/decoding of signals. The PHY layer (comprising PMD and PLCP) and the MAC layer will be explained in more detail in the following sections. Apart from the protocol sublayers, the standard specifies management layers and the station management. The **MAC management** supports the association and re-association of a station to an access point and roaming between different access points. It also controls authentication mechanisms, encryption, synchronization of a station with regard to an

access point, and power management to save battery power. MAC management also maintains the MAC management information base (MIB). The main tasks of the **PHY management** include channel tuning and PHY MIB maintenance. Finally, **station management** interacts with both management layers and is responsible for additional higher layer functions (e.g., control of bridging and interaction with the distribution system in the case of an access point).

Physical layer

IEEE 802.11 supports three different physical layers: one layer based on infra red and two layers based on radio transmission (primarily in the ISM band at 2.4 GHz, which is available worldwide). All PHY variants include the provision of the **clear channel assessment** signal **(CCA)**. This is needed for the MAC mechanisms controlling medium access and indicates if the medium is currently idle. The transmission technology (which will be discussed later) determines exactly how this signal is obtained. The PHY layer offers a service access point (SAP) with 1 or 2 Mbit/s transfer rate to the MAC layer (basic version of the standard). The remainder of this section presents the three versions of a PHY layer defined in the standard.

5. Explain the technique about Frequency hopping spread spectrum. (L-1,CO-1)

Frequency hopping spread spectrum (FHSS) is a spread spectrum technique which allows for the coexistence of multiple networks in the same area by separating different networks using different hopping sequences. The original standard defines 79 hopping channels for North America and Europe, and 23 hopping channels for Japan (each with a bandwidth of 1 MHz in the 2.4 GHz ISM band). The selection of a particular channel is achieved by using a pseudo-random hopping pattern. National restrictions also determine further parameters, e.g., maximum transmit power is 1 W in the US, 100 mW EIRP (equivalent isotropic radiated power) in Europe and 10 mW/MHz in Japan. The standard specifies Gaussian shaped FSK (frequency shift keying), GFSK, as modulation for the FHSS PHY. For 1 Mbit/s a 2 level GFSK is used (i.e., 1 bit is mapped to one frequency, a 4 level GFSK for 2 Mbit/s (i.e., 2 bits are mapped to one frequency).

While sending and receiving at 1 Mbit/s is mandatory for all devices, operation at 2 Mbit/s is optional. This facilitated the production of low-cost devices for the lower rate only and more powerful devices for both transmission rates in the early days of 802.11. Figure 7.7 shows a frame of the physical layer used with FHSS. The frame
consists of two basic parts, the PLCP part (preamble and header) and the payload part. While the PLCP part is always transmitted at 1 Mbit/s, payload, i.e. MAC data, can use 1 or 2 Mbit/s. Additionally, MAC data is scrambled using the polynomial s(z) = z7 + z4 + 1 for DC blocking and whitening of the spectrum.

The fields of the frame fulfill the following functions:

● **Synchronization:** The PLCP preamble starts with 80 bit synchronization, which is a 010101... bit pattern. This pattern is used for synchronization of potential receivers and signal detection by the CCA.

● **Start frame delimiter (SFD):** The following 16 bits indicate the start of the frame and provide frame synchronization. The SFD pattern is 0000110010111101.

● **PLCP_PDU length word (PLW):** This first field of the PLCP header indicates the length of the payload in bytes including the 32 bit CRC at the end of the payload. PLW can range between 0 and 4,095.

● **PLCP signalling field (PSF):** This 4 bit field indicates the data rate of the payload following. All bits set to zero (0000) indicates the lowest data rate of 1 Mbit/s. The granularity is 500 kbit/s, thus 2 Mbit/s is indicated by 0010 and the maximum is 8.5 Mbit/s (1111). This system obviously does not accommodate today's higher data rates.

● **Header error check (HEC)**: Finally, the PLCP header is protected by a 16 bit checksum with the standard ITU-T generator polynomial G(x) = x16 + x12 + x5 + 1.

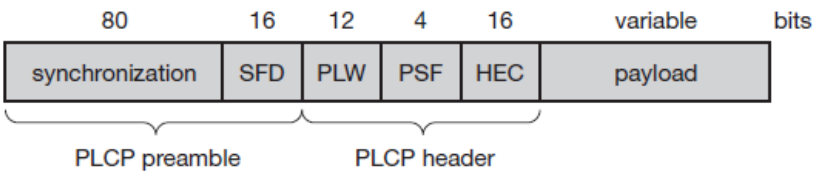


Figure 7.7 Format of an IEEE 802.11 PHY frame using FHSS

6. How the Direct sequence spread spectrum is utilized in WLAN techniques. (H-1,CO-1)

Direct sequence spread spectrum (DSSS) is the alternative spread spectrum method separating by code and not by frequency. In the case of IEEE 802.11 DSSS, spreading is achieved using the 11-chip Barker sequence (+1, –1, +1, +1, –1, +1, +1, +1, –1, –1, –1). The key characteristics of this method are its robustness against interference and its insensitivity to multipath propagation (time delay spread). However, the implementation is more complex compared to FHSS. IEEE 802.11 DSSS PHY also uses the 2.4 GHz ISM band and offers both 1 and 2 Mbit/s data rates. The system uses differential binary phase shift keying (DBPSK) for 1 Mbit/s transmission and differential quadrature phase shift keying (DQPSK) for 2 Mbit/s as modulation schemes. Again, the maximum transmit power is 1 W in the US, 100 mW EIRP in Europe and 10 mW/MHz in Japan. The symbol rate is 1 MHz, resulting in a chipping rate of 11 MHz. All bits transmitted by the DSSS PHY are scrambled with the polynomial s(z) = z7 + z4 + 1 for DC blocking and whitening of the spectrum. Many of today's products offering 11 Mbit/s according to 802.11b are still backward compatible to these lower data rates. Figure 7.8 shows a frame of the physical layer using DSSS. The frame consists of two basic parts, the PLCP part (preamble and header) and the payload part. While the PLCP part is always transmitted at 1 Mbit/s, payload, i.e., MAC
data, can use 1 or 2 Mbit/s. The fields of the frame have the following functions:

● **Synchronization:** The first 128 bits are not only used for synchronization, but also gain setting, energy detection (for the CCA), and frequency offset compensation. The synchronization field only consists of scrambled 1 bits.

● **Start frame delimiter (SFD):** This 16 bit field is used for synchronization at the beginning of a frame and consists of the pattern 1111001110100000.

● **Signal:** Originally, only two values have been defined for this field to indicate the data rate of the payload. The value 0x0A indicates 1 Mbit/s (and thus DBPSK), 0x14 indicates 2 Mbit/s (and thus DQPSK). Other values have been reserved for future use, i.e., higher bit rates. Coding for higher data rates is explained in sections 7.3.6 and 7.3.7.

● **Service:** This field is reserved for future use; however, 0x00 indicates an IEEE 802.11 compliant frame.

● **Length:** 16 bits are used in this case for length indication of the payload in microseconds.

● **Header error check (HEC):** Signal, service, and length fields are protected by this checksum using the ITU-T CRC-16 standard polynomial.

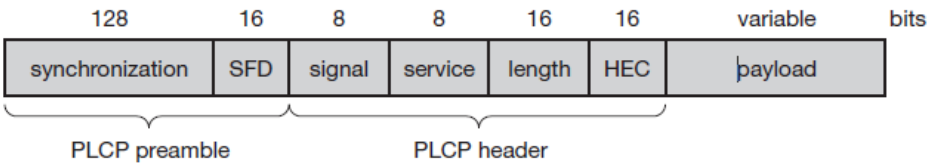


Figure 7.8 Format of an IEEE 802.11 PHY frame using DSSS

Infra red

The PHY layer, which is based on infra red (IR) transmission, uses near visible light at 850–950 nm. Infra red light is not regulated apart from safety restrictions (using lasers instead of LEDs). The standard does not require a line-of-sight between sender and receiver, but should also work with diffuse light. This allows for point-to-multipoint communication. The maximum range is about 10 m if no sunlight or heat sources interfere with the transmission. Typically, such a network will only work in buildings, e.g., classrooms, meeting rooms etc. Frequency reuse is very simple – a wall is more than

enough to shield one IR based IEEE 802.11 network from another. (comparison between IR and radio transmission and Wesel, 1998 for more details.) Today, no products are available that offer infra red communication based on 802.11. Proprietary products offer, e.g., up to 4 Mbit/s using diffuse infra red light. Alternatively, directed infra red communication based on IrDA can be used (IrDA, 2002).

Medium access control layer The MAC layer has to fulfill several tasks. First of all, it has to control medium access, but it can also offer support for roaming, authentication, and power conservation. The basic services provided by the MAC layer are the mandatory **asynchronous data service** and an optional **time-bounded service**. While 802.11 only offers the asynchronous service in ad-hoc network mode, both service types can be offered using an infrastructure-based network together with the access point coordinating medium access. The asynchronous service supports broadcast and multi-cast packets, and packet exchange is based on a 'best effort' model, i.e., no delay bounds can be given for transmission. The following three basic access mechanisms have been defined for IEEE 802.11: the mandatory basic method based on a version of CSMA/CA, an optional method avoiding the hidden terminal problem, and finally a contention- free polling method for time-bounded service. The first two methods are also summarized as **distributed coordination function (DCF)**, the third method is called **point coordination function (PCF**). DCF only offers asynchronous service, while PCF offers both asynchronous and time-bounded service but needs an access point to control medium access and to avoid contention. The MAC mechanisms are also called **distributed foundation wireless**

medium access control (DFWMAC). For all access methods, several parameters for controlling the waiting time before medium access are important. Figure 7.9 shows the three different rameters that define the priorities of medium access. The values of the parameters depend on the PHY and are defined in relation to a **slot** time. Slot time is derived from the medium propagation delay, transmitter delay, and other PHY dependent parameters. Slot time is 50 μs for FHSS and 20 μs for DSSS. The medium, as shown, can be busy or idle (which is detected by the CCA). If the medium is busy this can be due to data frames or other control frames. During a contention phase several nodes try to access the medium.

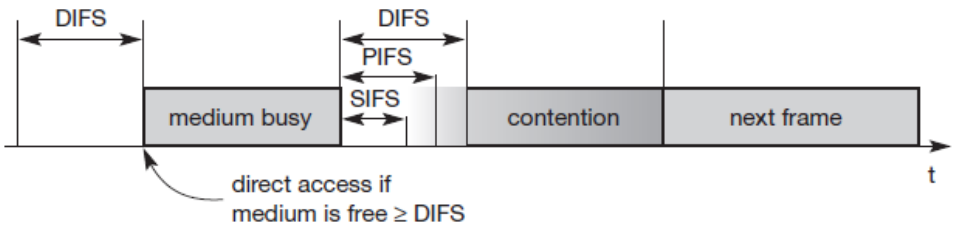


Figure 7.9
Medium access and inter-frame spacing

● **Short inter-frame spacing (SIFS):** The shortest waiting time for medium access (so the highest priority) is defined for short control messages, such as acknowledgements of data packets or polling responses. For DSSS SIFS is 10 *μ*s and for FHSS it is 28 *μ*s. The use of this parameter will be explained in sections 7.3.4.1 through 7.3.4.3.

● **PCF inter-frame spacing (PIFS):** A waiting time between DIFS and SIFS (and thus a medium priority) is used for a time-bounded service. An access point polling other nodes only has to wait PIFS for medium access (see section 7.3.4.3). PIFS is defined as SIFS plus one slot time.

● **DCF inter-frame spacing (DIFS):** This parameter denotes the longest waiting time and has the lowest priority for medium access. This waiting time is used for asynchronous data service within a contention period (this parameter and the basic access method are explained in section 7.3.4.1). DIFS is defined as SIFS plus two slot times. 7.3.4.1 Basic DFWMAC-DCF using CSMA/CA The mandatory access mechanism of IEEE 802.11 is based on **carrier sense multiple access with collision avoidance** (CSMA/CA), which is a random access scheme with carrier sense and collision avoidance through random backoff. The basic CSMA/CA mechanism is shown in Figure 7.10. If the medium is idle for at least the duration of DIFS (with the help of the CCA signal of the physical layer), a node can access the medium at once. This allows for short access delay under light load. But as more and more nodes try to access the medium, additional mechanisms are needed.

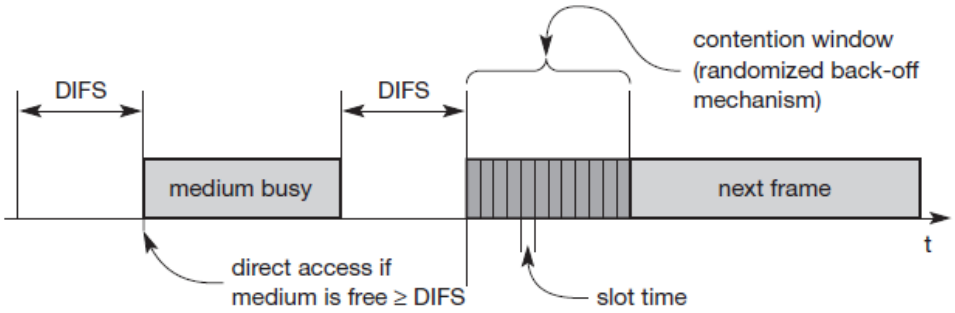


Figure 7.10
Contention window and waiting time

If the medium is busy, nodes have to wait for the duration of DIFS, entering a contention phase afterwards. Each node now chooses a **random backoff time** within a **contention window** and delays medium access for this random amount of time. The node continues to sense the medium. As soon as a node senses the channel is busy, it has lost this cycle and has to wait for the next

chance, i.e., until the medium is idle again for at least DIFS. But if the randomized additional waiting time for a node is over and the medium is still idle, the node can access the medium immediately (i.e., no other node has a shorter waiting time). The additional waiting time is measured in multiples of the above-mentioned slots. This additional randomly distributed delay helps to avoid collisions – otherwise all stations would try to transmit data after waiting

for the medium becoming idle again plus DIFS. Obviously, the basic CSMA/CA mechanism is not fair. Independent of the overall time a node has already waited for transmission; each node has the same chances for transmitting data in the next cycle. To provide fairness, IEEE 802.11

adds a **backoff timer.** Again, each node selects a random waiting time within the range of the contention window. If a certain station does not get access to the medium in the first cycle, it stops its backoff timer, waits for the channel to be idle again for DIFS and starts the counter again. As soon as the counter expires, the node accesses the medium. This means that deferred stations do not choose a randomized backoff time again, but continue to count down. Stations

that have waited longer have the advantage over stations that have just entered, in that they only have to wait for the remainder of their backoff timer from the previous cycle(s). Figure 7.11 explains the basic access mechanism of IEEE 802.11 for five stations trying

to send a packet at the marked points in time. Station3 has the first request from a higher layer to send a packet packet arrival at the MAC SAP). The station senses the medium, waits for DIFS and accesses the medium, i.e., sends the packet. Station1, station2, and station5 have to wait at least until the medium is idle for DIFS again after station3 has stopped sending. Now all three stations choose a backoff time within the contention window and start counting down their backoff timers.

Figure 7.11 shows the random backoff time of station1 as sum of boe (the elapsed backoff time) and bor (the residual backoff time). The same is shown for station5. Station2 has a total backoff time of only boe and gets access to the medium first. No residual backoff time for station2 is shown. The backoff timers of station1 and station5 stop, and the stations store their residual backoff times. While a new station has to choose its backoff time from the whole contention window, the two old stations have statistically smaller backoff values. The older values are on average lower than the new ones. Now station4 wants to send a packet as well, so after DIFS waiting time, three stations try to get access. It can now happen, as shown in the figure, that two stations accidentally have the same backoff time, no matter whether remaining or newly chosen. This results in a collision on the medium as shown, i.e., the trans-

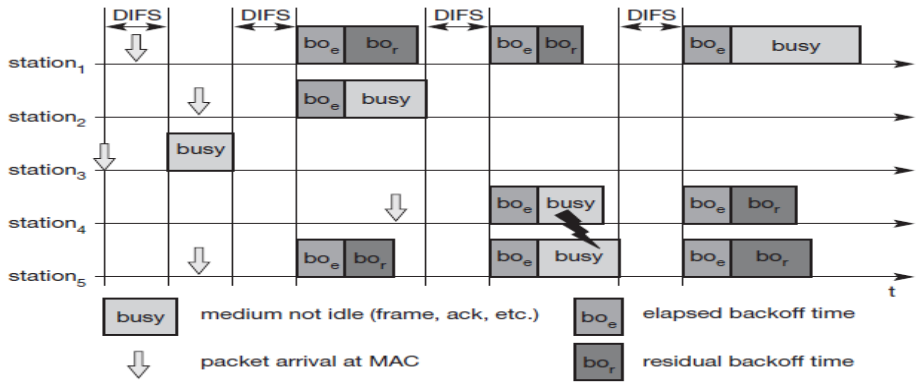
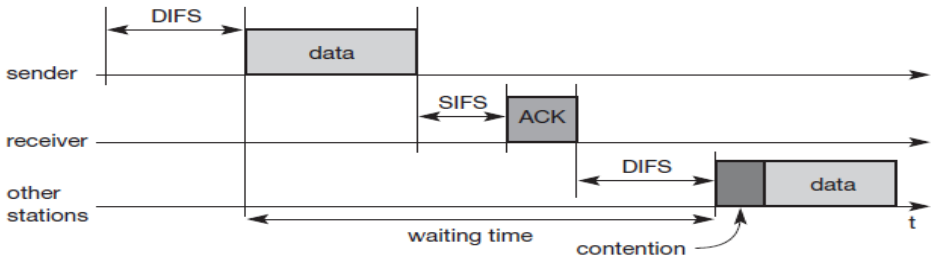


Figure 7.11 Basic DFWMAC–DCF with several competing senders

mitted frames are destroyed. Station1 stores its residual backoff time again. In the last cycle shown station1 finally gets access to the medium, while station4 and station5 have to wait. A collision triggers a retransmission with a new random selection of the backoff time. Retransmissions are not privileged. Still, the access scheme has problems under heavy or light load. Depending on the size of the contention window (CW), the random values can either be

too close together (causing too many collisions) or the values are too high (causing unnecessary delay). The system tries to adapt to the current number of stations trying to send. The contention window starts with a size of, e.g., CWmin = 7. Each time a collision occurs, indicating a higher load on the medium, the contention window doubles up to a maximum of, e.g., CWmax = 255 (the window can take on the values 7, 15, 31, 63, 127, and 255). The larger the contention window is, the greater is the resolution power of the randomized scheme. It is less likely to choose the same random backoff time using a large CW. However, under a light load, a small CW ensures shorter access delays. This algorithm is also called **exponential backoff** and is already familiar from IEEE 802.3 CSMA/CD in a similar version. While this process describes the complete access mechanism for broadcast frames, an additional feature is provided by the standard for unicast data transfer. Figure 7.12 shows a sender accessing the medium and sending its data. But now, the receiver answers directly with an **acknowledgement (ACK)**. The receiver accesses the medium after waiting for a duration of SIFS so no other station can access the medium in the meantime and cause a collision. The other stations have to wait for DIFS plus their backoff time. This acknowledgement ensures the correct reception (correct checksum CRC at the receiver) of a frame on the MAC layer, which is especially important in error-prone environments

Figure 7.12
IEEE 802.11 unicast data transfer



such as wireless connections. If no ACK is returned, the sender automatically retransmits the frame. But now the sender has to wait again and compete for the access right. There are no special rules for retransmissions. The number of retransmissions is limited, and final failure is reported to the higher layer. 7.3.4.2 DFWMAC-DCF with RTS/CTS extension Section 3.1 discussed the problem of hidden terminals, a situation that can also occur in IEEE 802.11 networks. This problem occurs if one station can receive two others, but those stations cannot receive each other. The two stations may

sense the channel is idle, send a frame, and cause a collision at the receiver in the middle. To deal with this problem, the standard defines an additional mechanism using two control packets, RTS and CTS. The use of the mechanism is optional; however, every 802.11 node has to implement the functions to react properly upon reception of RTS/CTS control packets. Figure 7.13 illustrates the use of RTS and CTS. After waiting for DIFS (plus a random backoff time if the medium was busy), the sender can issue a **request to send (RTS)** control packet. The RTS packet thus is not given any higher priority compared to other data packets. The RTS packet includes the receiver of the data transmission to come and the duration of the whole data transmission. This duration specifies the time interval necessary to transmit the whole data frame and the acknowledgement related to it. Every node receiving this RTS now has to set its **net allocation vector (NAV)** in accordance with the duration field. The NAV then specifies the earliest point at which the station can try to access the medium again. If the receiver of the data transmission receives the RTS, it answers with a **clear to send (CTS)** message after waiting for SIFS. This CTS packet contains the duration field again and all stations receiving this packet from the receiver of the intended data transmission have to adjust their NAV. The latter set of receivers need not be the same as the first set receiving the RTS packet. Now all nodes within receiving distance around sender and receiver are informed that they have to wait more time before accessing the medium. Basically, this mechanism reserves the medium for one sender exclusively (this is why it is sometimes called a virtual reservation scheme).

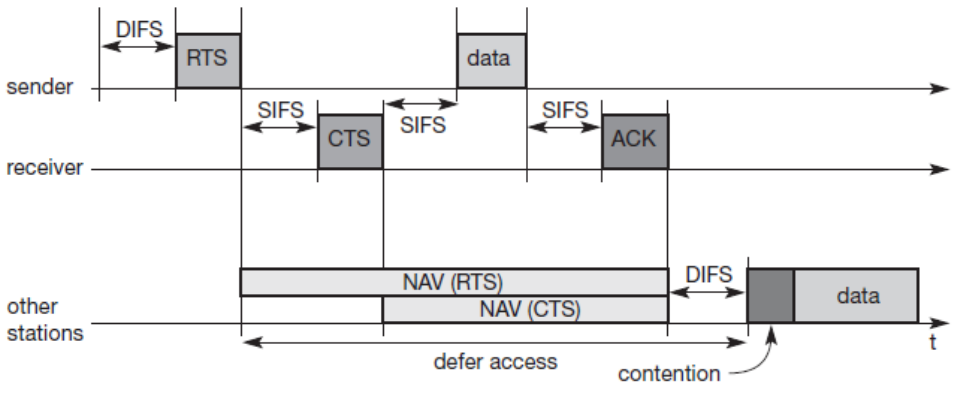


Figure 7.13
IEEE 802.11 hidden node provisions for contention-free access

Finally, the sender can send the data after SIFS. The receiver waits for SIFS after receiving the data packet and then acknowledges whether the transfer was correct. The transmission has now been completed, the NAV in each node marks the medium as free and the standard cycle can start again. Within this scenario (i.e., using RTS and CTS to avoid the hidden terminal problem), collisions can only occur at the beginning while the RTS is sent. Two or more stations may start sending at the same time (RTS or other data packets). Using RTS/CTS can result in a non-negligible overhead causing a waste of bandwidth and higher delay. An RTS threshold can determine when to use the additional mechanism (basically at larger frame sizes) and when to disable it (short frames). Chhaya (1996) and Chhaya (1997) give an overview of the synchronous services in 802.11 and discuss performance under different load scenarios. Wireless LANs have bit error rates in transmission that are typically several orders of magnitude higher than, e.g., fiber optics. The probability of an erroneous frame is much higher for wireless links assuming the same frame length. One way to decrease the error probability of frames is to use shorter frames. In this case, the bit error rate is the same, but now only short frames are destroyed and, the frame error rate decreases. However, the mechanism of fragmenting a user data packet into several smaller parts should be transparent for a user. The MAC layer should have the possibility of adjusting the transmission frame size to the current error rate on the medium. The IEEE 802.11 standard specifies a **fragmentation** mode (see Figure 7.14). Again, a sender can send an RTS control packet to reserve the medium after a waiting time of DIFS. This RTS packet now includes the duration for the transmission of the first fragment and the corresponding acknowledgement. A certain set of nodes may receive this RTS and set their NAV according to the duration field. The receiver answers with a CTS, again including the duration of the transmission up to the acknowledgement. A (possibly different) set of receivers gets this CTS message and sets the NAV.

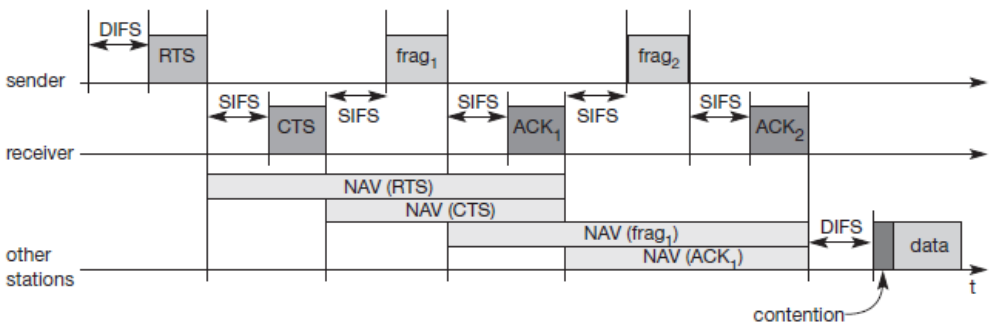


Figure 7.14 IEEE 802.11 fragmentation of user data

As shown in Figure 7.13, the sender can now send the first data frame, frag1, after waiting only for SIFS. The new aspect of this fragmentation mode is that it includes another duration value in the frame frag1. This duration field reserves the medium for the duration of the transmission following, comprising the second fragment and its acknowledgement. Again, several nodes may receive this reservation and adjust their NAV. If all nodes are static and transmission conditions have not changed, then the set of nodes receiving the duration field in frag1 should be the same as the set that has received the initial reservation in the RTS control packet. However, due to the mobility of nodes and changes in the environment, this could also be a different set of nodes.

The receiver of frag1 answers directly after SIFS with the acknowledgement packet ACK1 including the reservation for the next transmission as shown. Again, a fourth set of nodes may receive this reservation and adjust their NAV (which again could be the same as the second set of nodes that has received the reservation in the CTS frame). If frag2 was not the last frame of this transmission, it would also include a new duration for the third consecutive transmission. (In the example shown, frag2 is the last fragment of this transmission so the sender does not reserve the medium any longer.) The receiver acknowledges this second fragment, not reserving the medium again. After ACK2, all nodes can compete for the medium again after having waited for DIFS.

7. Explain the mechanisms for minimum transmission bandwidth . (H-1,CO-1)

The two access mechanisms presented so far cannot guarantee a maximum access delay or minimum transmission bandwidth. To provide a time-bounded service, the

standard specifies a **point coordination function (PCF)** on top of the standard DCF mechanisms. Using PCF requires an access point that controls medium access and polls the single nodes. Ad-hoc networks cannot use this function so, provide no QoS but 'best effort' in IEEE 802.11 WLANs.

The **point co-ordinator** in the access point splits the access time into super frame periods as shown in Figure 7.15. A **super frame** comprises a **contentionfree period** and a **contention period**. The contention period can be used for the two access mechanisms presented above. The figure also shows several wireless stations (all on the same line) and the stations' NAV (again on one line).

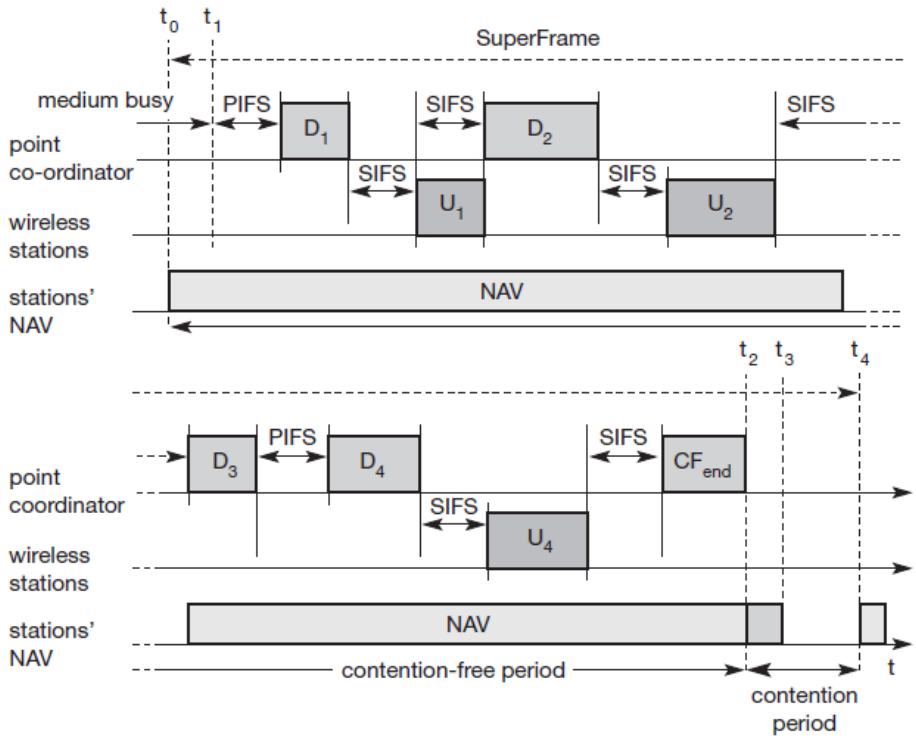


Figure 7.15 Contention-free access using polling mechanisms (PCF)

At time t0 the contention-free period of the super frame should theoretically start, but another station is still transmitting data (i.e., the medium is busy). This means that PCF also defers to DCF, and the start of the super frame may be postponed. The only possibility of avoiding variations is not to have any contention period at all. After the medium has been idle until t1, the point coordinator has to wait for PIFS before accessing the medium. As PIFS is smaller than DIFS, no other station can start sending earlier. The point coordinator now sends data D1 downstream to the first wireless station. This station can answer at once after SIFS (see Figure 7.15). After waiting for

SIFS again, the point coordinator can poll the second station by sending D2. This station may answer upstream to the coordinator with data U2. Polling continues with the third node. This time the node has nothing to answer and the point coordinator will not receive a packet after SIFS. After waiting for PIFS, the coordinator can resume polling the stations. Finally, the point coordinator can issue an end marker (CFend), indicating that the contention period may start again. Using PCF automatically sets the NAV, preventing other stations from sending. In the example, the contention-free period planned initially would have been from t0 to t3. However, the point coordinator finished polling earlier, shifting the end of the contention-free period to t2. At t4, the cycle starts again with the next super frame.

The transmission properties of the whole wireless network are now determined by the polling behavior of the access point. If only PCF is used and polling is distributed evenly, the bandwidth is also distributed evenly among all polled nodes. This would resemble a static, centrally controlled time division multiple access (TDMA) system with time division duplex (TDD) transmission. This method comes with an overhead if nodes have nothing to send, but the access point polls them permanently. Anastasi (1998) elaborates the example of voice transmission using 48 byte packets as payload. In this case, PCF introduces an overhead of 75 byte. 7.3.4.4 MAC frames Figure 7.16 shows the basic structure of an IEEE 802.11 MAC data frame together with the content of the frame control field. The fields in the figure refer to the following:

● **Frame control:** The first 2 bytes serve several purposes. They contain several sub-fields as explained after the MAC frame.

● **Duration/ID:** If the field value is less than 32,768, the duration field contains the value indicating the period of time in which the medium is occupied (in μs). This field is used for setting the NAV for the virtual reservation mechanism using RTS/CTS and during fragmentation. Certain values above 32,768 are reserved for identifiers.

● **Address 1 to 4:** The four address fields contain standard IEEE 802 MAC addresses (48 bit each), as they are known from other 802.x LANs. The meaning of each address

depends on the DS bits in the frame control field and is explained in more detail in a separate paragraph.

● **Sequence control:** Due to the acknowledgement mechanism frames may be duplicated. Therefore a sequence number is used to filter duplicates.

● **Data:** The MAC frame may contain arbitrary data (max. 2,312 byte), which is transferred transparently from a sender to the receiver(s).

● **Checksum (CRC):** Finally, a 32 bit checksum is used to protect the frame as it is common practice in all 802.x networks. The frame control field shown in Figure 7.16 contains the following fields:

● **Protocol version:** This 2 bit field indicates the current protocol version and is fixed to 0 by now. If major revisions to the standard make it incompatible with the current version, this value will be increased.

● **Type:** The type field determines the function of a frame: management (=00), control (=01), or data (=10). The value 11 is reserved. Each type has several subtypes as indicated in the following field.

● **Subtype:** Example subtypes for management frames are: 0000 for association request, 1000 for beacon. RTS is a control frame with subtype 1011, CTS is coded as 1100. User data is transmitted as data frame with subtype 0000. All details can be found in IEEE, 1999.

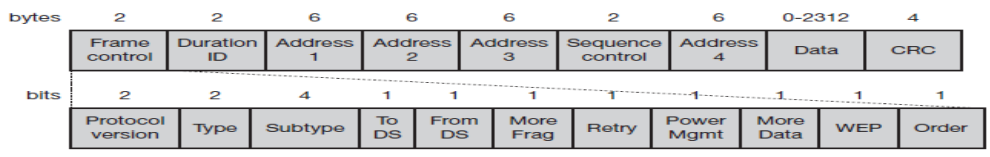


Figure 7.16
IEEE 802.11 MAC packet structure

● **To DS/From DS:** Explained in the following in more detail.

● **More fragments:** This field is set to 1 in all data or management frames that have another fragment of the current MSDU to follow.

● **Retry:** If the current frame is a retransmission of an earlier frame, this bit is set to 1. With the help of this bit it may be simpler for receivers to eliminate duplicate frames.

● **Power management:** This field indicates the mode of a station after successful transmission of a frame. Set to 1 the field indicates that the station goes into power-save mode. If the field is set to 0, the station stays active.

● **More data:** In general, this field is used to indicate a receiver that a sender has more data to send than the current frame. This can be used by an access point to indicate to a station in power-save mode that more packets are buffered. Or it can be used by a station to indicate to an access point after being polled that more polling is necessary as the station has more data ready to transmit.

● **Wired equivalent privacy (WEP):** This field indicates that the standard security mechanism of 802.11 is applied. However, due to many weaknesses found in the WEP algorithm higher layer security should be used to secure an 802.11 network (Borisov, 2001).

● **Order:** If this bit is set to 1 the received frames must be processed in strict order. MAC frames can be transmitted between mobile stations; between mobile stations and an access point and between access points over a DS (see Figure 7.3). Two bits within the Frame Control field, '**to DS**' and '**from DS**', differentiate these cases and control the meaning of the four addresses used. Table 7.1 gives an overview of the four possible bit values of the DS bits and the associated interpretation of the four address fields.

to DS	from DS	Address 1	Address 2	Address 3	Address 4
0	0	DA	SA	BSSID	–
0	1	DA	BSSID	SA	–
1	0	BSSID	SA	DA	–
1	1	RA	TA	DA	SA

Table 7.1 Interpretation of the MAC addresses in an 802.11 MAC frame

Every station, access point or wireless node, filters on **address 1**. This address identifies the physical receiver(s) of the frame. Based on this address, a station can decide whether the frame is relevant or not. The second address, **address 2**, represents the physical transmitter of a frame. This information is important because this particular sender is also the recipient of the MAC layer acknowledgement. If a packet from a transmitter (address 2) is received by the receiver with address 1, this receiver in turn acknowledges the data packet using address 2 as receiver address as shown in the ACK packet in Figure 7.17. The remaining two addresses, **address 3** and **address 4**, are mainly necessary for the logical assignment of frames (logical sender, BSS identifier, logical receiver). If address 4 is not needed the field is omitted. For addressing, the following four scenarios are possible:

● **Ad-hoc network:** If both DS bits are zero, the MAC frame constitutes a packet which is exchanged between two wireless nodes without a distribution system. **DA** indicates the **destination address**, **SA** the **source address** of the frame, which are identical to the physical receiver and sender addresses respectively. The third address identifies the **basic service set (BSSID)** (see Figure 7.4), the fourth address is unused.

● **Infrastructure network, from AP:** If only the 'from DS' bit is set, the frame physically originates from an access point. DA is the logical and physical receiver, the second address identifies the BSS, the third address specifies the logical sender, the source address of the MAC frame. This case is an example for a packet sent to the receiver via the access point.

● **Infrastructure network, to AP:** If a station sends a packet to another station via the access point, only the 'to DS' bit is set. Now the first address represents the physical receiver of the frame, the access point, via the BSS identifier. The second address is the logical and physical sender of the frame, while the third address indicates the logical receiver.

● **Infrastructure network, within DS:** For packets transmitted between two access points over the distribution system, both bits are set. The first **receiver address (RA)**, represents the MAC address of the receiving access point. Similarly, the second address **transmitter address (TA)**, identifies the sending access point within the distribution system. Now two more addresses are needed to identify the original destination DA of the frame and the original source of the frame SA. Without these additional addresses, some encapsulation mechanism would be necessary to transmit MAC frames over the distribution system transparently. Figure 7.17 shows three control packets as examples for many special packets defined in the standard. The **acknowledgement packet (ACK)** is used to acknowledge the correct reception of a data frame as shown in Figure 7.12. The receiver address is directly copied from the address 2 field of the immediately previous frame. If no more fragments follow for a certain frame the duration field is set to 0. Otherwise the duration value of the previous frame (minus the time required to transmit the ACK minus SIFS) is stored in the duration field.

8. Brief about MAC management techniques used in IEEE 802.11. (H-3,CO-1)

MAC management plays a central role in an IEEE 802.11 station as it more or less controls all functions related to system integration, i.e., integration of a wireless station into a BSS, formation of an ESS, synchronization of stations etc.

The following functional groups have been identified and will be discussed in more detail in the following sections:

● **Synchronization:** Functions to support finding a wireless LAN, synchronization of internal clocks, generation of beacon signals.

● **Power management:** Functions to control transmitter activity for power conservation, e.g., periodic sleep, buffering, without missing a frame.

● **Roaming:** Functions for joining a network (association), changing access points, scanning for access points.

● **Management information base (MIB):** All parameters representing the current state of a wireless station and an access point are stored within a MIB for internal and external access. A MIB can be accessed via standardized protocols such as the simple network management protocol (SNMP).

Synchronization

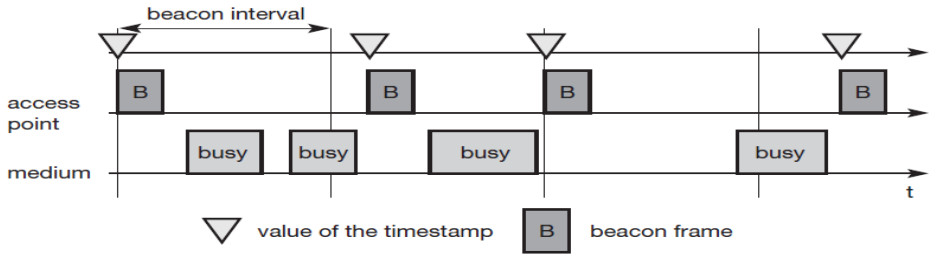
Each node of an 802.11 network maintains an internal clock. To synchronize the clocks of all nodes, IEEE 802.11 specifies a **timing synchronization function (TSF)**. As we will see in the following section, synchronized clocks are needed for power management, but also for coordination of the PCF and for synchronization of the hopping sequence in an FHSS system. Using PCF, the local timer of a node can predict the start of a super frame, i.e., the contention free and contention period. FHSS physical layers need the same hopping sequences so that all nodes can communicate within a BSS.

Within a BSS, timing is conveyed by the (quasi)periodic transmissions of a beacon frame. A **beacon** contains a timestamp and other management information used for power management and roaming (e.g., identification of the BSS). The timestamp is used by a node to adjust its local clock. The node is not required to hear every beacon to stay synchronized; however, from time to time internal clocks should be adjusted. The transmission of a beacon frame is not always periodic because the beacon frame is also deferred if the medium is busy. Within **infrastructure-based** networks, the access point performs synchronization by transmitting the (quasi)periodic beacon signal, whereas all other wireless nodes adjust their local timer to the time stamp. This represents the simple case shown in Figure 7.18. The access point is not always able to send its beacon B periodically if the medium is busy. However, the access point always tries to schedule transmissions according to the expected beacon interval (**target**

beacon transmission time), i.e., beacon intervals are not shifted if one beacon is delayed. The timestamp of a beacon always reflects the real transmit time, not the scheduled time.

For ad-hoc networks, the situation is slightly more complicated as they do not have an access point for beacon transmission. In this case, each node maintains its own synchronization timer and starts the transmission of a beacon frame after the beacon interval. Figure 7.19 shows an example where multiple stations try to send their beacon. However, the standard random backoff algorithm is also applied to the beacon frames so only one beacon wins. All other stations now adjust their internal clocks according to the received beacon and

Figure 7.18
Beacon transmission in a busy 802.11 infrastructure network



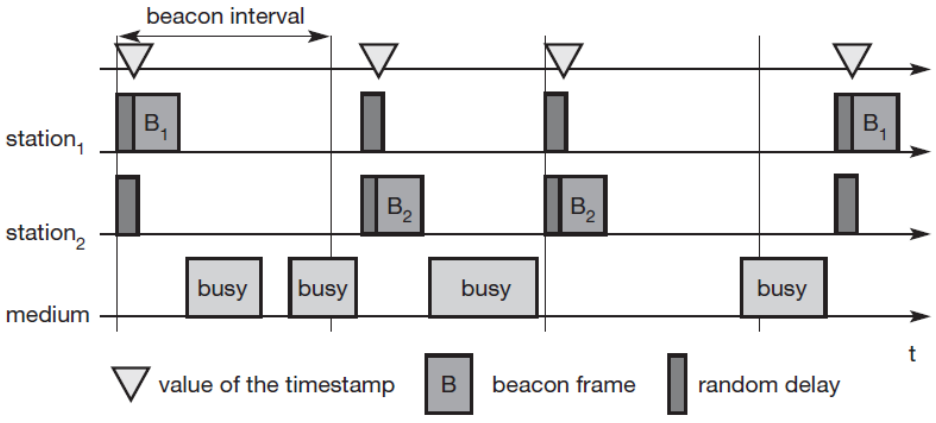


Figure 7.19
Beacon transmission in a busy 802.11 ad-hoc network

suppress their beacons for this cycle. If collision occurs, the beacon is lost. In this scenario, the beacon intervals can be shifted slightly because all clocks may vary as may the start of a beacon interval from a node's point of view. However, after successful synchronization all nodes again have the same consistent view.

Power management

Wireless devices are battery powered (unless a solar panel is used). Therefore, power-saving mechanisms are crucial for the commercial success of such devices. Standard LAN protocols assume that stations are always ready to receive data, although receivers are idle most of the time in lightly loaded networks. However, this permanent readiness of the receiving module is critical for battery life as the receiver current may be up to 100 mA (Woesner, 1998). The basic idea of IEEE 802.11 power management is to switch off the transceiver whenever it is not needed. For the sending device this is simple to achieve as the transfer is triggered by the device itself. However, since the power management of a receiver cannot know in advance when the transceiver has to be active for a specific packet, it has to 'wake up' the transceiver periodically.

Switching off the transceiver should be transparent to existing protocols and should be flexible enough to support different applications. However, throughput can be traded-off for battery life. Longer off-periods save battery life but reduce average throughput and vice versa. The basic idea of power saving includes two states for a station: **sleep** and **awake**, and buffering of data in senders. If a sender intends to communicate with a power-saving station it has to buffer data if the station is asleep. The sleeping station on the other hand has to wake up periodically and stay awake for a certain time. During this time, all senders can announce the destinations of their buffered data frames. If a station detects that it is a destination of a buffered packet it has to stay awake until the transmission takes place. Waking up at the right moment requires the **timing synchronization function (TSF)** introduced in section 7.3.5.1. All stations have to wake up or be awake at the same time.

Power management in **infrastructure**-based networks is much simpler compared to ad-hoc networks. The access point buffers all frames destined for stations

operating in power-save mode. With every beacon sent by the access point, a **traffic indication map (TIM)** is transmitted. The TIM contains a list of stations for which unicast data frames are buffered in the access point. The TSF assures that the sleeping stations will wake up periodically and listen to the beacon and TIM. If the TIM indicates a unicast frame buffered for the station, the station stays awake for transmission. For multi-cast/broadcast transmission, stations will always stay awake. Another reason for waking up is a frame which has to be transmitted from the station to the access point. A sleeping station still has the TSF timer running. Figure 7.20 shows an example with an access point and one station. The state of the medium is indicated. Again, the access point transmits a beacon frame each beacon interval. This interval is now the same as the TIM interval. Additionally, the access point maintains a **delivery traffic indication map (DTIM)** interval for sending broadcast/multicast frames. The DTIM interval is always a multiple of the TIM interval. All stations (in the example, only one is shown) wake up prior to an expected TIM or DTIM. In the first case, the access point has to transmit a broadcast frame and the station stays awake to receive it. After receiving the broadcast frame, the station returns to sleeping mode. The station wakes up again just before the next TIM transmission. This time the TIM is delayed due to a busy medium so, the station stays awake. The access point has nothing to send and the station goes back to sleep. At the next TIM interval, the access point indicates that the station is the destination for a buffered frame. The station answers with a **PS** (power saving) **poll** and stays awake to receive data. The access point then transmits the data for the station, the station acknowledges the receipt and may also send some

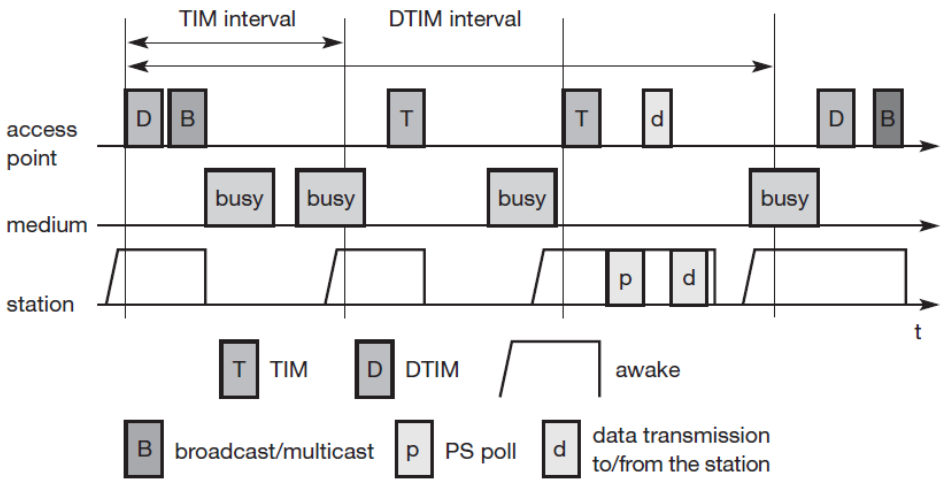


Figure 7.20 Power management in IEEE 802.11 infrastructure networks

data (as shown in the example). This is acknowledged by the access point (acknowledgments are not shown in the figure). Afterwards, the station switches to sleep mode again. Finally, the access point has more broadcast data to send at the next DTIM interval, which is again deferred by a busy medium. Depending on internal thresholds, a station may stay awake if the sleeping period would be too short. This mechanism clearly shows the trade-off between short delays in station access and saving battery power. The shorter the TIM interval, the shorter the delay, but the lower the power-saving effect. In ad-hoc networks, power management is much more complicated than in infrastructure networks. In this case, there is no access point to buffer data in

one location but each station needs the ability to buffer data if it wants to communicate with a power-saving station. All stations now announce a list of buffered frames during a period when they are all awake. Destinations are announced using **ad-hoc traffic indication map (ATIMs)** – the announcement period is called the **ATIM window**. Figure 7.21 shows a simple ad-hoc network with two stations. Again, the beacon interval is determined by a distributed function (different stations may send the beacon). However, due to this synchronization, all stations within the ad-hoc network wake up at the same time. All stations stay awake for the ATIM interval as shown in the first two steps and go to sleep again if no frame is buffered for them. In the third step, station1 has data buffered for station2. This is indicated in an ATIM transmitted by station1.

Station2 acknowledges this ATIM and stays awake for the transmission. After the ATIM window, station1 can transmit the data frame, and station2 acknowledges its receipt. In this case, the stations stay awake for the next beacon.

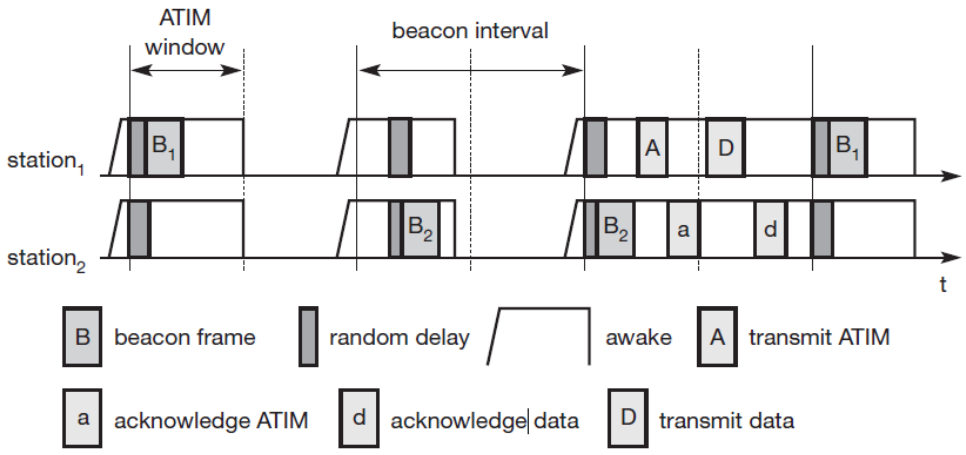


Figure 7.21 Power management in IEEE 802.11 ad-hoc networks

One problem with this approach is that of scale. If many stations within an ad-hoc network operate in power-save mode, they may also want to transmit their ATIM within the ATIM window. More ATIM transmissions take place, more collisions happen and more stations are deferred. The access delay of large networks is difficult to predict. QoS guarantees can not be given under heavy load.

8. Explain the clear concept about roaming. (H-1,CO-1)

Typically, wireless networks within buildings require more than just one access point to cover all rooms. Depending on the solidity and material of the walls, one access point has a transmission range of 10–20 m if transmission is to be of decent quality. Each storey of a building needs its own access point(s) as quite often walls are thinner than floors. If a user walks around with a wireless station, the station has to move from one access point to another to provide uninterrupted service. Moving between access points is called **roaming**. The term “handover” or “handoff” as used in the context of mobile or cellular phone systems would be more appropriate as it is simply a change of the active cell. However, for WLANs roaming is more common.

The steps for roaming between access points are:

● A station decides that the current link quality to its access point AP1 is too poor. The station then starts **scanning** for another access point.

● Scanning involves the active search for another BSS and can also be used for setting up a new BSS in case of ad-hoc networks. IEEE 802.11 specifies scanning on single or multiple channels (if available at the physical layer) and differentiates between passive scanning and active scanning. **Passive scanning** simply means listening into the medium to find other networks, i.e.,
receiving the beacon of another network issued by the synchronization function within an access point. **Active scanning** comprises sending a **probe** on each channel and waiting for a response. Beacon and probe responses contain the information necessary to join the new BSS.

● The station then selects the best access point for roaming based on, e.g., signal strength, and sends an **association request** to the selected access point AP2.

● The new access point AP2 answers with an **association response**. If the response is successful, the station has roamed to the new access point AP2. Otherwise, the station has to continue scanning for new access points.

● The access point accepting an association request indicates the new station in its BSS to the distribution system (DS). The DS then updates its database, which contains the current location of the wireless stations. This database is needed for forwarding frames between different BSSs, i.e. between the different access points controlling the BSSs, which combine to form an ESS (see Figure 7.3). Additionally, the DS can inform the old access point AP1 that the station is no longer within its BSS.

Unfortunately, many products implemented proprietary or incompatible versions of protocols that support roaming and inform the old access point about the change in the

station's location. The standard **IEEE 802.11f (Inter Access Point Protocol, IAPP)** should provide a compatible solution for all vendors. This also includes load-balancing between access points and key generation for security algorithms based on IEEE 802.1x (IEEE, 2001).

9. Explain in detail about 802.11b standard. (L-1,CO-1)

As standardization took some time, the capabilities of the physical layers also evolved. Soon after the first commercial 802.11 products came on the market some companies offered proprietary solutions with 11 Mbit/s. To avoid market segmentation, a common standard, **IEEE 802.11b** (IEEE 1999) soon followed and was added as supplement to the original standard (Higher-speed physical layer extension in the 2.4 GHz band). This standard describes a new PHY layer and is by far the most successful version of IEEE 802.11 available today. Do not get confused about the fact that 802.11b hit the market before 802.11a. The standards are named according to the order in which the respective study groups have been established. As the name of the supplement implies, this standard only defines a new PHY layer. All the MAC schemes, management procedures etc. explained above are still used. Depending on the current interference and the distance between sender and receiver 802.11b systems offer 11, 5.5, 2, or 1 Mbit/s. Maximum user data rate is approx 6 Mbit/s. The lower data rates 1 and 2 Mbit/s use the 11-chip Barker sequence as explained in section 7.3.3.2 and DBPSK or DQPSK, respectively.

The new data rates, 5.5 and 11 Mbit/s, use 8-chip **complementary code keying (CCK)** (see IEEE, 1999, or Pahlavan, 2002, for details). The standard defines several packet formats for the physical layer. The mandatory format interoperates with the original versions of 802.11. The

optional versions provide a more efficient data transfer due to shorter headers/different coding schemes and can coexist with other 802.11 versions. However, the standard states that control all frames shall be transmitted at one of the basic rates, so they will

be understood by all stations in a BSS. Figure 7.22 shows two packet formats standardized for 802.11b. The mandatory format is called **long PLCP PPDU** and is similar to the format illustrated in Figure 7.8. One difference is the rate encoded in the signal field this is encoded in multiples of 100 kbit/s. Thus, 0x0A represents 1 Mbit/s, 0x14 is used for 2 Mbit/s, 0x37 for 5.5 Mbit/s and 0x6E for 11 Mbit/s. Note that the preamble and the header are transmitted at 1 Mbit/s using DBPSK. The optional **short PLCP PPDU** format differs in several ways. The short synchronization field consists of 56 scrambled zeros instead of scrambled ones. The short start frame delimiter SFD consists of a mirrored bit pattern compared to the SFD of the long format: 0000 0101 1100 1111 is used for the short PLCP PDU instead of 1111 0011 1010 0000 for the long PLCP PPDU. Receivers that are unable to receive the short format will not detect the start of a frame (but will sense the medium

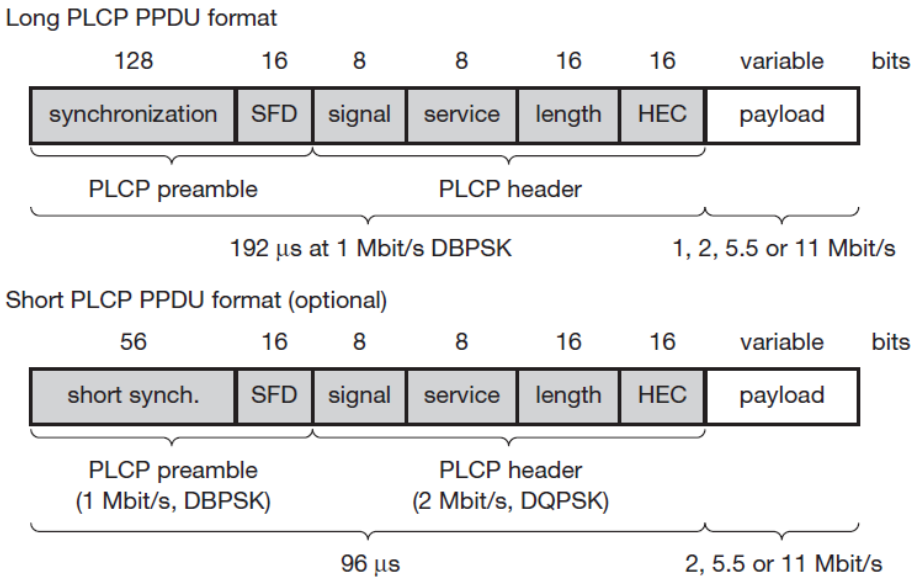


Figure 7.22 IEEE 802.11b PHY packet formats

is busy). Only the preamble is transmitted at 1 Mbit/s, DBPSK. The following header is already transmitted at 2 Mbit/s, DQPSK, which is also the lowest available data rate. As Figure 7.22 shows, the length of the overhead is only half for the short frames (96 μs instead of 192 μs). This is useful for, e.g., short, but timecritical, data transmissions. As IEEE 802.11b is the most widespread version, some more information is given for practical usage. The standards operates (like the DSSS version of 802.11) on certain

frequencies in the 2.4 GHz ISM band. These depend on national regulations. Altogether 14 channels have been defined as Table 7.2 shows. For each channel the center frequency is given. Depending on national restrictions 11 US/Canada), 13 (Europe with some exceptions) or 14 channels (Japan) can be used. Figure 7.23 illustrates the non-overlapping usage of channels for an IEEE 802.11b installation with minimal interference in the US/Canada and Europe. The spacing between the center frequencies should be at least 25 MHz (the occupied bandwidth of the main lobe of the signal is 22 MHz). This results in the channels 1, 6, and 11 for the US/Canada or 1, 7, 13 for Europe, respectively. It may be the case that, e.g., travellers from the US cannot use the additional channels (12 and 13) in Europe as their hardware is limited to 11 channels. Some European installations use channel 13 to minimize interference. Users can install overlapping cells for WLANs using the three non-overlapping channels to provide seamless coverage. This is similar to the cell planning for
mobile phone systems.

Table 7.2 Channel plan for IEEE 802.11b

Channel	Frequency [MHz]	US/Canada	Europe	Japan
1	2412	X	X	X
2	2417	X	X	X
3	2422	X	X	X
4	2427	X	X	X
5	2432	X	X	X
6	2437	X	X	X
7	2442	X	X	X
8	2447	X	X	X
9	2452	X	X	X
10	2457	X	X	X
11	2462	X	X	X
12	2467	–	X	X
13	2472	–	X	X
14	2484	–	–	X

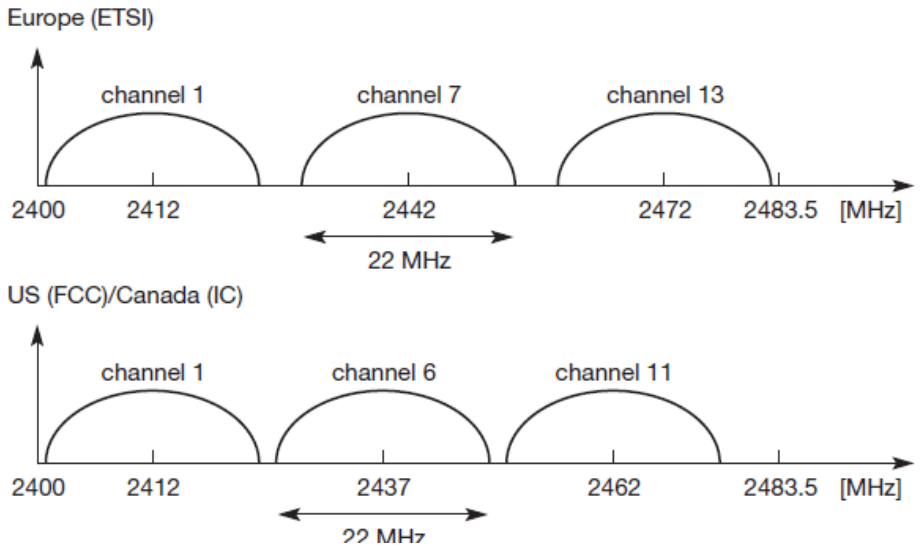


Figure 7.23
IEEE 802.11b non-overlapping channel selection

10.Briefly explain concepts used in 802.11a standard. (L-1,CO-1)

Initially aimed at the US 5 GHz U-NII (Unlicensed National Information Infrastructure) bands **IEEE 802.11a** offers up to 54 Mbit/s using OFDM (IEEE, 1999). The first products were available in 2001 and can now be used (after some harmonization between IEEE and ETSI) in Europe. The FCC (US) regulations offer three different 100 MHz domains for the use of 802.11a, each with a different legal maximum power output: 5.15–5.25 GHz/50 mW, 5.25–5.35 GHz/250 mW, and 5.725–5.825 GHz/1 W. ETSI (Europe) defines different frequency bands for

Europe: 5.15–5.35 GHz and 5.47–5.725 GHz and requires two additional mechanisms for operation: dynamic frequency selection (DFS) and transmit power control (TPC) which will be explained in the context of HiperLAN2 in more detail. (This is also the reason for introducing IEEE 802.11h, see section 7.3.8.) Maximum transmit power is 200 mW EIRP for the lower frequency band (indoor use) and 1 W EIRP for the higher frequency band (indoor and outdoor use). DFS and TPC are not necessary, if the transmit power stays below 50 mW EIRP and only 5.15–5.25 GHz are used. Japan allows operation in the frequency range 5.15–5.25 GHz and requires carrier sensing every 4 ms to minimize interference. Up to now, only 100 MHz are available ‘worldwide’

at 5.15–5.25 GHz. The physical layer of IEEE 802.11a and the ETSI standard HiperLAN2 has been jointly developed, so both physical layers are almost identical. Most statements and explanations in the following, which are related to the transmission technology are also valid for HiperLAN2. However, HiperLAN2 differs in the MAC layer, the PHY layer packet formats, and the offered services (quality of service, real time etc.). This is discussed in more detail in section 7.4. It should be noted that most of the development for the physical layer for 802.11a was adopted from the HiperLAN2 standardization – but 802.11a products were available first and are already in widespread use. Again, IEEE 802.11a uses the same MAC layer as all 802.11 physical layers do and, in the following, only the lowest layer is explained in some detail. To be able to offer data rates up to 54 Mbit/s IEEE 802.11a uses many different technologies. The system uses 52 subcarriers (48 data + 4 pilot) that are modulated using BPSK, QPSK, 16-QAM, or 64-QAM. To mitigate transmission errors, FEC is applied using coding rates of 1/2, 2/3, or 3/4. Table 7.3 gives an overview of the standardized combinations of modulation and coding schemes together with the resulting data rates. To offer a data rate of 12 Mbit/s, 96 bits are coded into one OFDM symbol. These 96 bits are distributed over 48 subcarriers and 2 bits are modulated per sub-carrier using QPSK (2 bits per point in the constellation diagram). Using a coding rate of 1/2 only 48 data bits can be transmitted.

Data rate [Mbit/s]	Modulation	Coding rate	Coded bits per subcarrier	Coded bits per OFDM symbol	Data bits per OFDM symbol
6	BPSK	1/2	1	48	24
9	BPSK	3/4	1	48	36
12	QPSK	1/2	2	96	48
18	QPSK	3/4	2	96	72
24	16-QAM	1/2	4	192	96
36	16-QAM	3/4	4	192	144
48	64-QAM	2/3	6	288	192
54	64-QAM	3/4	6	288	216

Table 7.3 Rate dependent parameters for IEEE 802.11a

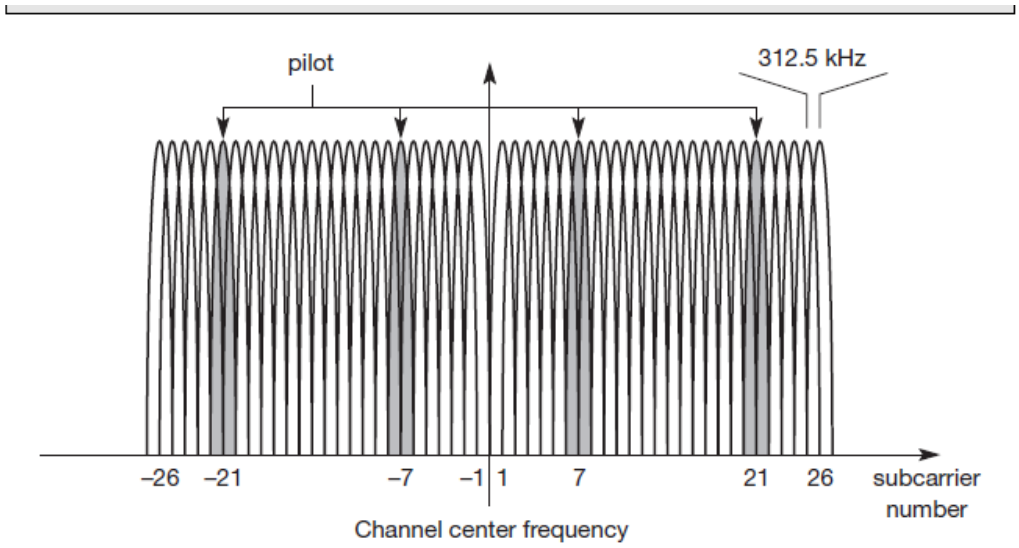


Figure 7.24
Usage of OFDM in IEEE 802.11a

Figure 7.24 shows the usage of OFDM in IEEE 802.11a. Remember, the basic idea of OFDM (or MCM in general) was the reduction of the symbol rate by distributing bits over numerous subcarriers. IEEE 802.11a uses a fixed symbol rate of 250,000 symbols per second independent of the data rate (0.8 μs guard interval for ISI mitigation plus 3.2 μs used for data results in a symbol duration of 4 μs). As Figure 7.24 shows, 52 subcarriers are equally spaced around a center frequency. (Center frequencies will be explained later). The spacing between the subcarriers is 312.5 kHz. 26 subcarriers are to the left of the center frequency and 26 are to the right. The center frequency itself is not used as subcarrier. Subcarriers with the numbers –21, –7, 7, and 21 are used for pilot signals to make the signal detection robust against frequency offsets.

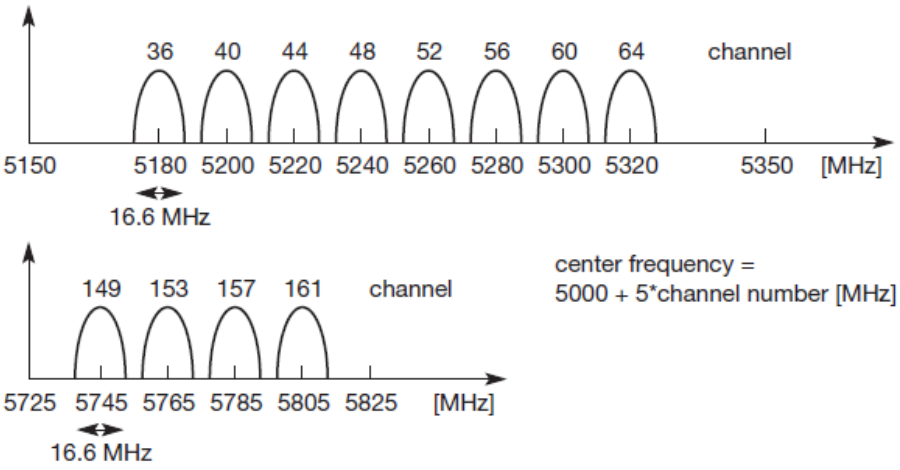


Figure 7.25 Operating channels of IEEE 802.11a in the U-NII bands

Similar to 802.11b several operating channels have been standardized to minimize interference. Figure 7.25 shows the **channel layout** for the US U-NII bands. The center frequency of a channel is 5000 + 5*channel number [MHz]. This definition provides a unique numbering of channels with 5 MHz spacing starting from 5 GHz. Depending on national regulations, different sets of channels may be used. Eight channels have been defined for the lower two bands in the U-NII (36, 40, 44, 48, 52, 56, 60, and 64); four more are available in the high band (149, 153, 157, and 161). Using these channels allows for interference-free operation of overlapping 802.11a cells. Channel spacing is 20 MHz, the occupied bandwidth of 802.11a is 16.6 MHz. How is this related to the spacing of the sub-carriers? 20 MHz/64 equals 312.5 kHz. 802.11a uses 48 carriers for data, 4 for pilot signals, and 12 carriers are sometimes called virtual subcarriers. (Set to zero, they do not contribute to the data transmission but may be used for an implementation of OFDM with the help of FFT, see IEEE, 1999, or ETSI, 2001a, for more details). Multiplying 312.5 kHz by 52 subcarriers and adding the extra space for the center frequency results in approximately 16.6 MHz occupied bandwidth per channel (details of the transmit spectral power mask neglected, see ETSI, 2001a). Due to the nature of OFDM, the PDU on the physical layer of IEEE 802.11a looks quite different from 802.11b or the original 802.11 physical layers. Figure 7.26 shows the basic structure of an **IEEE 802.11a PPDU**.

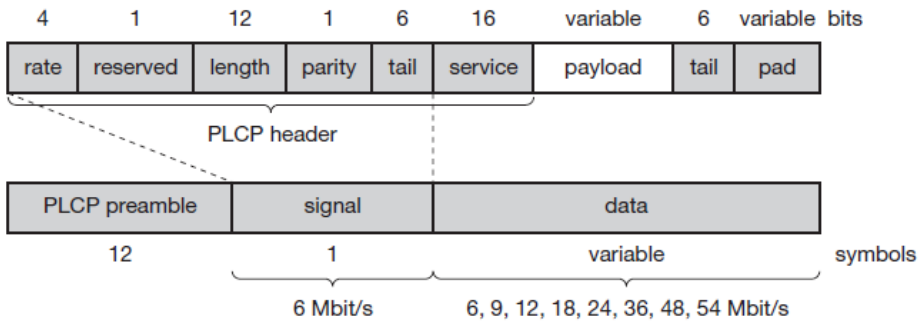


Figure 7.26
IEEE 802.11a physical layer PDU

The **PLCP preamble** consists of 12 symbols and is used for frequency acquisition, channel estimation, and synchronization. The duration of the preamble is 16 μs.

● The following OFDM symbol, called **signal**, contains the following fields and is BPSK-modulated. The 4 bit **rate** field determines the data rate and the modulation of the rest of the packet (examples are 0x3 for 54 Mbit/s, 0x9 for 24 Mbit/s, or 0xF for 9 Mbit/s). The **length** field indicates the number of bytes in the payload field. The **parity** bit shall be an even parity for the first 16 bits of the signal field (rate, length and the reserved bit). Finally, the six **tail** bits are set to zero.

● The **data** field is sent with the rate determined in the rate field and contains a **service** field which is used to synchronize the descrambler of the receiver (the data stream is scrambled using the polynomial x7 + x4 + 1) and which contains bits for future use. The **payload** contains the MAC PDU (1-4095 byte). The **tail** bits are used to reset the encoder. Finally, the **pad** field ensures that the number of bits in the PDU maps to an integer number of OFDM symbols.

Compared to IEEE 802.11b working at 2.4 GHz IEEE 802.11a at 5 GHz offers much higher data rates. However, shading at 5 GHz is much more severe compared to 2.4 GHz and depending on the SNR, propagation conditions and the distance between sender and receiver, data rates may drop fast (e.g., 54 Mbit/s may be available only in an LOS or near LOS condition). Additionally, the MAC layer of IEEE 802.11 adds overheads. User data rates are therefore much lower than the data rates listed above. Typical user rates in Mbit/s are (transmission rates in brackets) 5.3 (6), 18 (24), 24 (36), and 32 (54). The following section presents some additional developments in the

context of 802.11, which also comprise a standard for higher data rates at 2.4 GHz that can benefit from the better propagation conditions at lower frequencies.

12. Explain the some of the Newer developments in other IEEE standard protocols.

(H-1,CO-1)

While many products that follow the IEEE 802.11a and 802.11b standards are available, several new groups have been formed within the IEEE to discuss enhancements of the standard and new applications. As things change fast, the current status can be checked via (IEEE, 2002a). The following is only a selection of ongoing work (at the time of writing). The completed standards **IEEE 802.11c** and **802.11d** cover additions for bridging support and updates for physical layer
requirements in different regulatory domains (i.e., countries).

● **802.11e (MAC enhancements):** Currently, the 802.11 standards offer no quality of service in the DCF operation mode. Some QoS guarantees can be given, only via polling using PCF. For applications such as audio, video, or media stream, distribution service classes have to be provided. For this reason, the MAC layer must be enhanced compared to the current standard.

● **802.11f (Inter-Access Point Protocol):** The current standard only describes the basic architecture of 802.11 networks and their components. The implementation of components, such as the distribution system, was deliberately not specified. Specifications of implementations should generally be avoided as they hinder improvements. However, a great flexibility in the implementation combined with a lack of detailed interface definitions and communication protocols, e.g., for management severely limits the interoperability of devices from different vendors. For example, seamless roaming between access points of different vendors is often impossible. 802.11f standardizes the necessary exchange of information between access points to support the functions of a distribution system.

● **802.11g (Data rates above 20 Mbit/s at 2.4 GHz):** Introducing new modulation schemes, forward error correction and OFDM also allows for higher data rates at 2.4 GHz. This approach should be backward compatible to 802.11b and should benefit from the better propagation characteristics at 2.4 GHz compared to 5 GHz. Currently, chips for 54 Mbit/s are available as well as first products. An alternative (or additional) proposal for 802.11g suggests the so called packet binary convolutional coding (PBCC) to reach a data rate of 22 Mbit/s (Heegard, 2001). While the 54 Mbit/s OFDM mode is mandatory, the 22 Mbit/s PBCC mode can be used as an option. The decision between 802.11a and 802.11g is not obvious. Many 802.11a products are already available and the 5 GHz band is (currently) not as crowded as the 2.4 GHz band where not only microwave ovens, but also Bluetooth, operate (see section 7.5). Coverage is better at 2.4 GHz and fewer access points are needed, lowering the overall system cost. 802.11g access points can also communicate with 802.11b devices as the current 802.11g products show. Dual mode (or then triple mode) devices will be available covering 802.11a and b (and g). If a high traffic volume per square meter is expected (e.g., hot spots in airport terminals), the smaller cells of 802.11a access points and the higher number of available channels (to avoid interference) at 5 GHz are clear advantages.

802.11h (Spectrum managed 802.11a): The 802.11a standard was primarily designed for usage in the US U-NII bands. The standardization did not consider non-US regulations such as the European requirements for power control and dynamic selection of the transmit frequency. To enable the regulatory acceptance of 5 GHz products, dynamic channel selection (DCS) and transmit power control (TPC) mechanisms (as also specified for the European HiperLAN2 standard) have been added. With this extension, 802.11a products can also be operated in Europe. These additional mechanisms try to balance the load in the 5 GHz band.

● **802.11i (Enhanced Security mechanisms):** As the original security mechanisms (WEP) proved to be too weak soon after the deployment of the first products (Borisov, 2001), this working group discusses stronger encryption and authentication mechanisms. IEEE 802.1x will play a majorrole in this process.

Additionally, IEEE 802.11 has several **study groups** for new and upcoming topics. The group 'Radio Resource Measurements' investigates the possibilities of 802.11 devices to provide measurements of radio resources. Solutions for even higher throughput are discussed in the 'High Throughput' study group. Both groups had their first meetings in 2002. The first study group recently became the IEEE project 802.11k 'Radio Resource Measurement Enhancements.'

13. Evolution of HIPERLAAN and other improvements – Explain in detail (H-3,CO-1)

In 1996, the ETSI standardized HIPERLAN 1 as a WLAN allowing for node mobility and supporting ad-hoc and infrastructure-based topologies (ETSI, 1996). (HIPERLAN stands for **high performance local area network**.) **HIPERLAN 1** was originally one out of four HIPERLANs envisaged, as ETSI decided to have different types of networks for different purposes. The key feature of all four networks is their integration of time-sensitive data transfer services. Over time, names have changed and the former HIPERLANs 2, 3, and 4 are now
called HiperLAN2, HIPERACCESS, and HIPERLINK. The current focus is on HiperLAN2, a standard that comprises many elements from ETSI's **BRAN** (broadband radio access networks) and **wireless ATM** activities. Neither wireless ATM nor HIPERLAN 1 were a commercial success. However, the standardization efforts had a lot of impact on QoS supporting wireless broadband networks such as **HiperLAN2**. Before describing HiperLAN2 in more detail, the following three sections explain key features of, and the motivation behind, HIPERLAN 1, wireless ATM, and BRAN.

Readers not interested in the historical background may proceed directly to section 7.4.4.

Historical: HIPERLAN 1

ETSI (1998b) describes HIPERLAN 1 as a wireless LAN supporting priorities and packet life time for data transfer at 23.5 Mbit/s, including forwarding mechanisms, topology discovery, user data encryption, network identification and power conservation mechanisms. HIPERLAN 1 should operate at 5.1–5.3 GHz with a range of 50 m in buildings at 1 W transmit power. The service offered by a HIPERLAN 1 is compatible with the standard MAC services known from IEEE 802.x LANs. Addressing is based on standard 48 bit MAC addresses. A special HIPERLAN 1 identification scheme allows the concurrent operation of two or more physically overlapping HIPERLANs without mingling their communication. Confidentiality is ensured by an encryption/decryption algorithm that requires the identical keys and initialization vectors for successful decryption of a data stream encrypted by a sender.

An innovative feature of HIPERLAN 1, which many other wireless networks do not offer, is its ability to forward data packets using several relays. Relays can extend the communication on the MAC layer beyond the radio range. For power conservation, a node may set up a specific wake-up pattern. This pattern determines at what time the node is ready to receive, so that at other times, the node can turn off its receiver and save energy. These nodes are called p-savers and need so-called p-supporters that contain information about the wake-up patterns of all the p-savers they are responsible for. A p-supporter only forwards data to a p-saver at the moment the p-saver is awake. This action also requires buffering mechanisms for packets on p-supporting forwarders. The following describes only the medium access scheme of HIPERLAN 1, a scheme that provides QoS and a powerful prioritization scheme. However, it turned out that priorities and QoS in general are not that important for standard LAN applications today. IEEE 802.11 in its standard versions does not offer priorities, the optional PCF is typically not implemented in products – yet 802.11 is very popular.

Elimination-yield non-preemptive priority multiple access (EY-NPMA) is not only a complex acronym, but also the heart of the channel access providing priorities and different access schemes. EY-NPMA divides the medium access of different competing nodes into three phases:

● **Prioritization:** Determine the highest priority of a data packet ready to be sent by competing nodes.

● **Contention:** Eliminate all but one of the contenders, if more than one sender has the highest current priority.

● **Transmission:** Finally, transmit the packet of the remaining node.

In a case where several nodes compete for the medium, all three phases are necessary (called 'channel access in **synchronized channel condition**'). If the channel is free for at least 2,000 so-called high rate bit-periods plus a dynamic extension, only the third phase, i.e. transmission, is needed (called 'channel

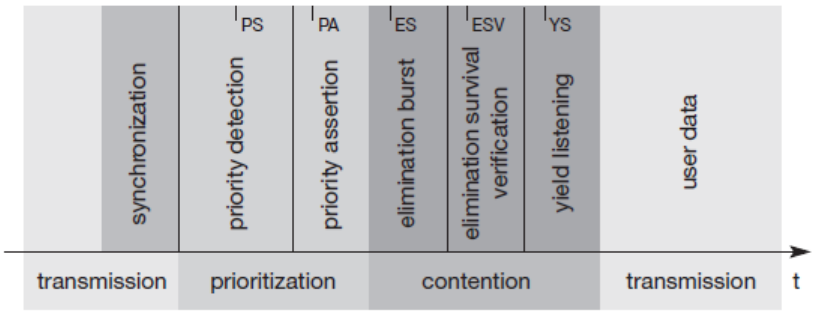


Figure 7.27 Phases of the HIPERLAN 1 EY-NPMA access scheme

access in **channel-free condition**'). The dynamic extension is randomly chosen between 0 and 3 times 200 high rate bit-periods with equal likelihood. This extension further minimizes the probability of collisions accessing a free channel if stations are synchronized on higher layers and try to access the free channel at the same time. HIPERLAN 1 also supports 'channel access in the **hidden elimination condition**' to handle the problem of hidden terminals as described in ETSI (1998b).

The contention phase is further subdivided into an **elimination phase** and a **yield phase**. The purpose of the elimination phase is to eliminate as many contending nodes as possible (but surely not all). The result of the elimination phase is a more or less constant number of remaining nodes, almost independent of the initial number of competing nodes. Finally, the yield phase completes the work of the elimination phase

with the goal of only one remaining node. Figure 7.27 gives an overview of the three main phases and some more details which will be explained in the following sections. For every node ready to send data, the access cycle starts with synchronization to the current sender. The first phase, prioritization, follows. After that, the elimination and yield part of the contention phase follow. Finally, the remaining node can transmit its data. Every phase has a certain duration which is measured in numbers of slots and is determined by the variables IPS, IPA, IES, IESV, and IYS.

Prioritization phase

HIPERLAN 1 offers five different priorities for data packets ready to be sent. After one node has finished sending, many other nodes can compete for the right to send. The first objective of the prioritization phase is to make sure that no node with a lower priority gains access to the medium while packets with higher priority are waiting at other nodes. This mechanism always grants nodes with higher priority access to the medium, no matter how high the load on lower priorities.

In the first step of the prioritization phase, the priority detection, time is divided into five slots, slot 0 (highest priority) to slot 4 (lowest priority). Each slot has a duration of IPS = 168 high rate bit-periods. If a node has the access priority p, it has to listen into the medium for p slots (priority detection). If the node senses the medium is idle for the whole period of p slots, the node asserts

the priority by immediately transmitting a burst for the duration IPA = 168 high rate bit-periods (priority assertion). The burst consists of the following high rate bit sequence, which is repeated as many times as necessary for the duration of the burst:1111101010001001110000011001011O

If the node senses activity in the medium, it stops its attempt to send data in this transmission cycle and waits for the next one. The whole prioritization phase ends as soon as one node asserts the access priority with a burst. This means that the prioritization phase is not limited by a fixed length, but depends on the highest priority.

Let us assume, for example, that there are three nodes with data ready to be sent, the packets of node 1 and node 2 having the priority 2, the packet of node 3 having the

priority 4. Then nodes 1, 2 and 3 listen into the medium and sense slots 0 and 1 are idle. Nodes 1 and 2 both send a burst in slot 2 as priority assertion.

Node 3 stops its attempt to transmit its packet. In this example, the prioritization phase has taken three slots.After this first phase at least one of the contending nodes will survive, the surviving nodes being all nodes with the highest priority of this cycle.

Elimination phase

Several nodes may now enter the elimination phase. Again, time is divided into slots, using the elimination slot interval IES = 212 high rate bit periods. The length of an individual elimination burst is 0 to 12 slot intervals long, the probability of bursting within a slot is 0.5. The probability PE(n) of an elimination burst to be n elimination slot intervals long is given by:

- PE(n) = 0.5n+1 for $0 \leq n < 12$
- PE(n) = 0.512 for n = 12

The elimination phase now resolves contention by means of elimination bursting and elimination survival verification. Each contending node sends an elimination burst with length n as determined via the probabilities and then listens to the channel during the survival verification interval IESV = 256 high rate bit periods. The burst sent is the same as for the priority assertion. A contending node survives this elimination phase if, and only if, it senses the channel is idle during its survival verification period. Otherwise, the node is eliminated and stops its attempt to send data during this transmission cycle. The whole elimination phase will last for the duration of the longest elimination burst among the contending nodes plus the survival verification time.

One or more nodes will survive this elimination phase, and can then continue with the next phase.

Yield phase

During the yield phase, the remaining nodes only listen into the medium without sending any additional bursts. Again, time is divided into slots, this time called yield slots with a duration of IYS = 168 high rate bit-periods. The length of an individual yield listening period can be 0 to 9 slots with equal likelihood. The probability PY(n) for a yield listening period to be n slots long is 0.1 for all n, $0 \leq n \leq 9$. Each node now listens for its yield listening period. If it senses the channel is idle during the whole period, it has survived

the yield listening. Otherwise, it withdraws for the rest of the current transmission cycle. This time, the length of the yield phase is determined by the shortest yield-listening period among all the contending nodes. At least one node will survive this phase and can start to transmit data. This is what the other nodes with longer yield listening period can sense. It is important to note that at this point there can still be more than one surviving node so a collision is still possible.

Transmission phase

A node that has survived the prioritization and contention phase can now send its data, called a low bit-rate high bit-rate HIPERLAN 1 CAC protocol data unit (LBR-HBR HCPDU). This PDU can either be multicast or unicast. In case of a unicast transmission, the sender expects to receive an immediate acknowledgement from the destination, called an acknowledgement HCPDU (AK-HCPDU), which is an LBR HCPDU containing only an LBR part.

Quality of service support and other specialties

The specialty of HIPERLAN 1 is its QoS support. The quality of service offered by the MAC layer is based on three parameters (**HMQoS-parameters**). The user can set a priority for data, priority = 0 denotes a high priority, priority = 1, a low priority. The user can determine the lifetime of an MSDU to specify time bounded delivery. The **MSDU lifetime** specifies the maximum time that can elapse between sending and receiving an MSDU. Beyond this, delivery of the MSDU becomes unnecessary. The MSDU lifetime has a range of 0–16,000 ms.

The **residual MSDU lifetime** shows the remaining lifetime of a packet. Besides data transfer, the MAC layer offers functions for looking up other HIPERLANs within radio range as well as special power conserving functions. **Power conservation** is achieved by setting up certain recurring patterns when a node can receive data instead of constantly being ready to receive. Special group-attendance patterns can be defined to enable multicasting. All nodes participating in a multicast group must be ready to receive at the same time when a sender transmits data.

HIPERLAN 1 MAC also offers user data **encryption** and **decryption** using a simple XOR-scheme together with random numbers. A key is chosen from a set of keys using a key identifier (KID) and is used together with an initialization vector IV to initialize the

pseudo random number generator. This random sequence is XORed with the user data (UD) to generate the encrypted data. Decryption of the encrypted UD works the same way, using the same random number sequence. This is not a strong encryption scheme – encryption is left to higher layers.

Table 7.4 Mapping of the normalized residual lifetime to the CAC priority

NRL	MSDU priority = 0	MSDU priority = 1
NRL < 10 ms	0	1
10 ms ≤ NRL < 20 ms	1	2
20 ms ≤ NRL < 40 ms	2	3
40 ms ≤ NRL < 80 ms	3	4
80 ms ≤ NRL	4	4

- **Link supervision:** LMP has to control the activity of a link, it may set up new SCO links, or it may declare the failure of a link.

● **State and transmission mode change:** Devices might switch the master/slave role, detach themselves from a connection, or change the operating mode. The available modes will be explained together with Figure 7.51. With transmission power of up to 100 mW, Bluetooth devices can have a range of up to 100 m. Having this power and relying on batteries, a Bluetooth device cannot be in an active transmit mode all the time. Bluetooth defines several low-power states for a device. Figure 7.51 shows the major states of a Bluetooth device and typical transitions. Every device, which is currently not participating in a piconet (and not switched off), is in **standby** mode. This is a low-power mode where only the native clock is running. The next step towards the **inquiry** mode can happen in two different ways. Either a device wants to establish a piconet or a device just wants to listen to see if something is going on.

● A device wants to establish a piconet: A user of the device wants to scan for other devices in the radio range. The device starts the inquiry procedure by sending an inquiry access code (IAC) that is common to all Bluetooth devices. The IAC is broadcast over 32 so-called wake-up carriers in turn.

● Devices in standby that listen periodically: Devices in standby may enter the inquiry mode periodically to search for IAC messages on the wake-up carriers. As soon as a

device detects an inquiry it returns a packet containing its device address and timing information required by the master to initiate a connection. From that moment on, the device acts as slave.If the inquiry was successful, a device enters the page mode. The inquiry phase is not coordinated; inquiry messages and answers to these messages may collide, so it may take a while before the inquiry is successful. After a while (typically seconds but sometimes up to a minute) a Bluetooth device sees all the devices in its radio range. During the **page** state two different roles are defined. After finding all required devices the master is able to set up connections to each device, i.e., setting up a piconet. Depending on the device addresses received the master calculates special hopping sequences to contact each device individually. The slaves answer and synchronize with the master's clock, i.e., start with the hopping sequence defined by the master. The master may continue to page more devices that will be added to the piconet. As soon as a device synchronizes to the hopping pattern of the piconet it also enters the connection state.The connection state comprises the active state and the low power states park, sniff, and hold. In the **active** state the slave participates in the piconet by listening, transmitting, and receiving. ACL and SCO links can be used. A master periodically synchronizes with all slaves. All devices being active must have the 3-bit **active member address** (AMA). Within the active state devices either transmit data or are simply connected. A device can enter standby again, via a detach procedure.

To save battery power, a Bluetooth device can go into one of three low power states:

● **Sniff state:** The sniff state has the highest power consumption of the low power states. Here, the device listens to the piconet at a reduced rate (not on every other slot as is the case in the active state). The interval for listening into the medium can be programed and is application dependent. The master designates a reduced number of slots for transmission to slaves in sniff state. However, the device keeps its AMA.

● **Hold state:** The device does not release its AMA but stops ACL transmission. A slave may still exchange SCO packets. If there is no activity in the piconet, the slave may either reduce power consumption or participate in another piconet.

● **Park state:** In this state the device has the lowest duty cycle and the lowest power consumption. The device releases its AMA and receives a parked member address (PMA). The device is still a member of the piconet, but gives room for another device to become active (AMA is only 3 bit, PMA 8 bit).Parked devices are still FH synchronized and wake up at certain beacon intervals for re-synchronization. All PDUs sent to parked slaves are broadcast.

Operating mode	Average current [mA]
SCO, HV1	53
SCO, HV3, 1 s interval sniff mode	26
ACL, 723.2 kbit/s	53
ACL, 115.2 kbit/s	15.5
ACL, 38.4 kbit/s, 40 ms interval sniff mode	4
ACL, 38.4 kbit/s, 1.28 s interval sniff mode	0.5
Park mode, 1.28 s beacon interval	0.6
Standby (no RF activity)	0.047

Table 7.7 Example power consumption (CSR, 2002)

The effect of the low power states is shown in Table 7.7. This table shows the typical average power consumption of a Bluetooth device (BlueCore2, CSR, 2002). It is obvious that higher data rates also require more transmission power. The intervals in sniff mode also influence power consumption. Typical IEEE 802.11b products have an average current in the order of 200 mA while receiving, 300 mA while sending, and 20 mA in standby.

15.How the L2CAP is used in WLAN applications?

The **logical link control and adaptation protocol (L2CAP)** is a data link control protocol on top of the baseband layer offering logical channels between Bluetooth devices with QoS properties. L2CAP is available for ACLs only. Audio applications using SCOs have to use the baseband layer directly (see Figure 7.44). L2CAP provides three different types of logical channels that are transported via the ACL between master and slave:

- **Connectionless:** These unidirectional channels are typically used for broadcasts from a master to its slave(s).
- **Connection-oriented:** Each channel of this type is bi-directional and supports QoS flow specifications for each direction. These flow specs follow RFC 1363 (Partridge, 1992) and define average/peak data rate, maximum burst size, latency, and jitter.
- **Signaling:** This third type of logical channel is used to exchanging signaling messages between L2CAP entities.

Each channel can be identified by its **channel identifier (CID)**. Signaling channels always use a CID value of 1, a CID value of 2 is reserved for connectionless channels. For connection-oriented channels a unique CID (>= 64) is dynamically assigned at each end of the channel to identify the connection

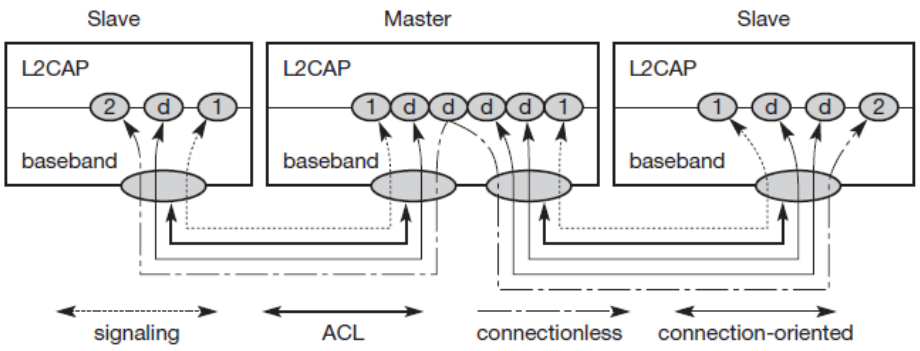


Figure 7.52 Logical channels between devices

(CIDs 3 to 63 are reserved). Figure 7.52 gives an example for logical channels using the ACL link between master and slave. The master has a bi-directional signaling channel to each slave. The CID at each end is 1. Additionally, the master maintains a connectionless, unidirectional channel to both slaves. The CID at the slaves is 2, while the CID at the beginning of the connectionless channel is dynamically assigned.

L2CAP provides mechanisms to add slaves to, and remove slaves from, such a multicast group. The master has one connection oriented channel to the left slave and two to the right slave. All CIDs for these channels are dynamically assigned (between 64 and 65535).

Figure 7.53 shows the three packet types belonging to the three logical channel types. The **length** field indicates the length of the payload (plus PSM for connectionless PDUs). The **CID** has the multiplexing/demultiplexing function as explained above. For connectionless PDUs a **protocol/service multiplexor (PSM)** field is needed to identify the higher layer recipient for the payload. For connection-oriented PDUs the CID already fulfills this function. Several PSM values have been defined, e.g., 1 (SDP), 3 (RFCOMM), 5 (TCS-BIN).Values above 4096 can be assigned dynamically. The payload of the signaling PDU contains one or more **commands**. Each command has its own **code** (e.g.,for command reject, connection request, disconnection response etc.) and an **ID** that matches a request with its reply. The **length** field indicates the length of the **data** field for this command.

Besides protocol multiplexing, flow specification, and group management, the L2CAP layer also provides segmentation and reassembly functions. Depending on the baseband capabilities, large packets have to be chopped into smaller segments. DH5 links, for example, can carry a maximum of 339 bytes while the L2CAP layer accepts up to 64 kbyte.

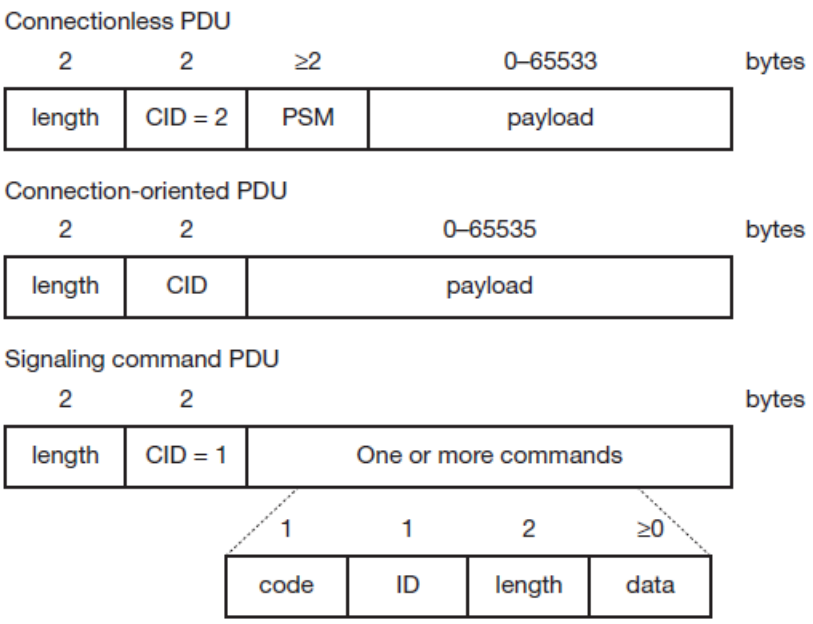


Figure 7.53 L2CAP packet formats

Security

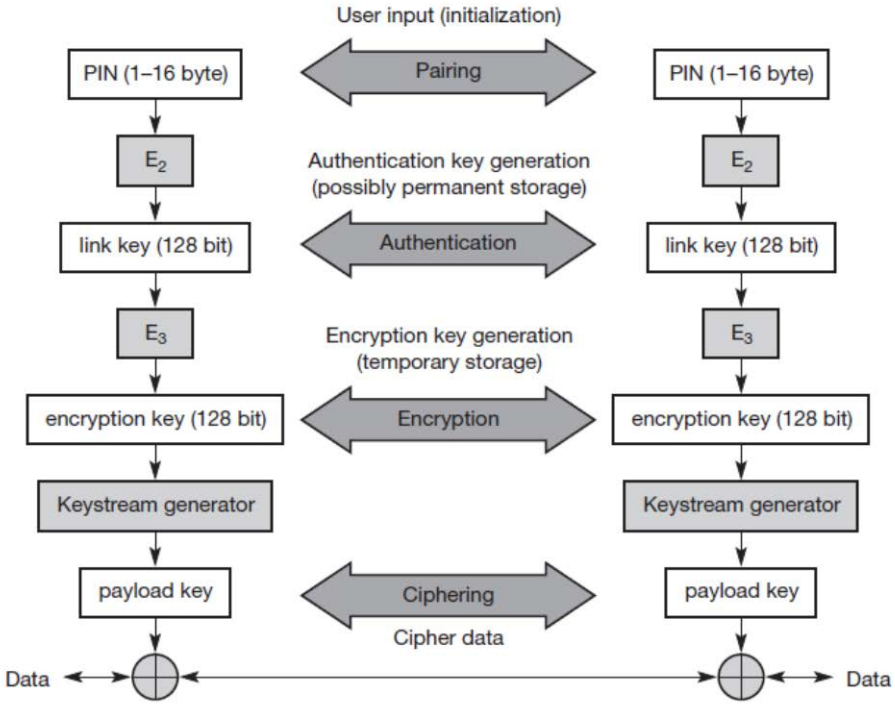
A radio interface is by nature easy to access. Bluetooth devices can transmit private data, e.g., schedules between a PDA and a mobile phone. A user clearlydoes not want another person to eavesdrop the data transfer. Just imagine a scenariowhere two Bluetooth enabled PDAs in suitcases 'meet' on the conveyor belt of an airport exchanging personal information! Bluetooth offers mechanisms for authentication and encryption on the MAC layer, which must be implemented in the same way within each device.

The main security features offered by Bluetooth include a challenge response routine for authentication, a stream cipher for encryption, and a session key generation. Each connection may require a one-way, two-way, or no authentication using the challenge-response routine. All these schemes have to be implemented in silicon, and higher layers should offer stronger encryption if needed. The security features included in Bluetooth only help to set up a local domain of trust between devices.

The security algorithms use the public identity of a device, a secret private user key, and an internally generated random key as input parameters. For each transaction, a new random number is generated on the Bluetooth chip. Key management is left to higher layer software.

Figure 7.54 shows several steps in the security architecture of Bluetooth. The illustration is simplified and the interested reader is referred to Bluetooth (2001a) for further details. The first step, called **pairing**, is necessary if two Bluetooth devices have never met before. To set up trust between the two devices a user can enter a secret PIN into both devices. This PIN can have a length of up to 16 byte. Unfortunately, most devices limit the length to four digits or, even worse, program

Figure 7.54
Bluetooth security components and protocols



the devices with the fixed PIN '0000' rendering the whole security concept of Bluetooth questionable at least. Based on the PIN, the device address, and random numbers, several keys can be computed which can be used as link key for **authentication**. Link keys are typically stored in a persistent storage. The authentication is a challenge-response process based on the link key, a random number generated by a verifier (the device that requests authentication), and the device address of the claimant (the device that is authenticated). Based on the link key, values generated during the authentication, and again a random number an encryption key is generated during the **encryption** stage of the security architecture. This key has a maximum size of 128 bits

and can be individually generated for each transmission. Based on the encryption key, the device address and the current clock a payload key is generated for ciphering user data. The payload key is a stream of pseudo-random bits. The **ciphering** process is a simple XOR of the user data and the payload key. Compared to WEP in 802.11, Bluetooth offers a lot more security. However, Bluetooth, too, has some weaknesses when it comes to real implementations.

The PINs are quite often fixed. Some of the keys are permanently stored on the devices and the quality of the random number generators has not been specified. If Bluetooth devices are switched on they can be detected unless they operate in the non-discoverable mode (no answers to inquiry requests). Either a user can use all services as intended by the Bluetooth system, or the devices are hidden to protect privacy. Either roaming profiles can be established, or devices

are hidden and, thus many services will not work. If a lot of people carry Bluetooth devices (mobile phones, PDAs etc.) this could give, e.g., department stores, a lot of information regarding consumer behavior.

16.How the SDP techniques is used so far I WLANs? (H-1,CO-1)

Bluetooth devices should work together with other devices in unknown environments in an ad-hoc fashion. It is essential to know what devices, or more specifically what services, are available in radio proximity. To find new services,Bluetooth defined the **service discovery protocol (SDP)**. SDP defines only the discovery of services, not their usage. Discovered services can be cached and gradual discovery is possible. Devices that want to offer a service have to install an SDP server. For all other devices an SDP client is sufficient. All the information an SDP server has about a service is contained in a **service record**. This consists of a list of service attributes and is identified by a 32-bit service record handle. SDP does not inform clients of any added or removed services. There is no service access control or service brokerage. A **service attribute** consists of an attribute ID and an attribute value. The 16-bit **attribute ID** distinguishes each service attribute from other service attributes within a service record. The attribute ID also identifies the semantics of the associated attribute value. The

attribute value can be an integer, a UUID (universally unique identifier), a string, a Boolean, a URL (uniform resource locator) etc. Table 7.8 gives some example attributes. The service handle as well as the ID list must be present. The ID list contains the UUIDs of the service classes in increasing generality (from the specific color postscript printer to

Attribute name	Attribute ID	Attribute value type	Example
ServiceRecordHandle	0000	32-bit unsigned integer	1f3e4723
ServiceClassIDList	0001	Data element sequence (UUIDs)	ColorPostscriptPrinterService ClassID, PostscriptPrinterService ClassID, PrinterServiceClassID
ProtocolDescriptorList	0004	Data element sequence	((L2CAP, PSM=RFCOMM), (RFCOMM, CN=2), (PPP), (IP), (TCP), (IPP))
DocumentationURL	000A	URL	www.xy.zz/print/srvs.html
IconURL	000C	URL	www.xy.zz/print/ico.png
ServiceName	0100	String	Color Printer

Table 7.8 Example attributes for an SDP service record

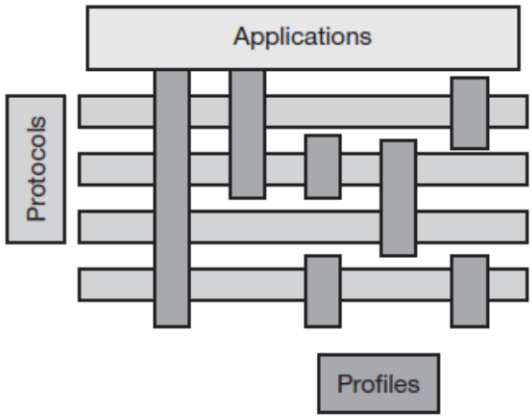
printers in general). The protocol descriptor list comprises the protocols needed to access this service. Additionally, the URLs for service documentation, an icon for the service and a service name which can be displayed together with the icon are stored in the example service record.

Profiles

Although Bluetooth started as a very simple architecture for spontaneous ad-hoc communication, many different protocols, components, extensions, and mechanisms have been developed over the last years. Application designers and vendors can implement similar, or even identical, services in many different ways using different components and protocols from the Bluetooth core standard. To provide compatibility among the devices offering the same services,Bluetooth specified many profiles in addition to the core protocols. Without the profiles too many parameters in Bluetooth would make interoperation between devices from different manufacturers almost impossible. **Profiles** represent default solutions for a certain usage model. They use a

selection of protocols and parameter set to form a basis for interoperability. Protocols can be seen as horizontal layers while profiles are vertical slices (as illustrated in Figure 7.55). The following **basic profiles** have been specified: generic access, service discovery, cordless telephony, intercom, serial port, headset, dialup networking, fax, LAN access, generic object exchange, object push, file transfer, and synchronization. **Additional profiles** are: advanced audio distribution, PAN, audio video remote control, basic printing, basic imaging, extended service discovery, generic audio video distribution, hands-free, and hardcopy cable replacement. Each profile selects a set of protocols. For example, the serial port profile needs RFCOMM, SDP, LMP, L2CAP. Baseband and radio are always required. The profile further defines all interoperability requirements, such as RS232 control signals for RFCOMM or configuration options for L2CAP (QoS, max. transmission unit).

Figure 7.55
Bluetooth profiles



IEEE 802.15

In 1999 the IEEE established a working group for wireless personal area networks (WPAN) with similar goals to Bluetooth. The working group was divided into several subgroups focusing on different aspects of WPANs (IEEE, 2002c).

The following gives a quick overview and presents the standard for low-rate WPANs, 802.15.4, in some more detail:

● **IEEE 802.15.1:** This group standardizes the lower layers of **Bluetooth** together with the Bluetooth consortium. IEEE LANs focus only on the physical and data link layer,

while the Bluetooth standard also comprises higher layers, application profiles, service description etc. as explained above.

● **IEEE 802.15.2:** The **coexistence** of wireless personal area networks (WPAN) and wireless local area networks (WLAN) is the focus of this group. One task is to quantify mutual interference and to develop algorithms and protocols for coexistence. Without additional mechanisms, Bluetooth/802.15.1 may act like a rogue member of an IEEE 802.11 network. Bluetooth is not aware of gaps, inter-frame spacing, frame structures etc. Figure 7.56 illustrates the problem. As explained in section 7.3, WLANs following the IEEE 802.11b standard may use three non-overlapping channels that are chosen during installation of the access points. Bluetooth/802.15.1 networks use a frequency hopping pattern to separate different piconets – 79 channels can be used. Without additional mechanisms, the hopping pattern of Bluetooth is independent of 802.11b's channel selection. Both systems work in the 2.4 GHz ISM band and might interfere with each other. Figure 7.56 shows two hopping sequences of two piconets interfering with several data packets, acknowledgements, and inter-frame spacings of 802.11b. The real effects of the interference range from 'almost no effect' to 'complete breakdown of the WLAN'. Publications on this issue differ depending on the test scenario,traffic load, signal power, propagation conditions etc. (Lansford, 2001). However, it seems that Bluetooth with its FHSS scheme is more robust than 802.11b with CSMA/CA (Pahlavan, 2002). To overcome the interference problems between 802.11b and 802.15.1, however severe they might be,the 802.15.2 working group proposes **adaptive frequency hopping**.

This

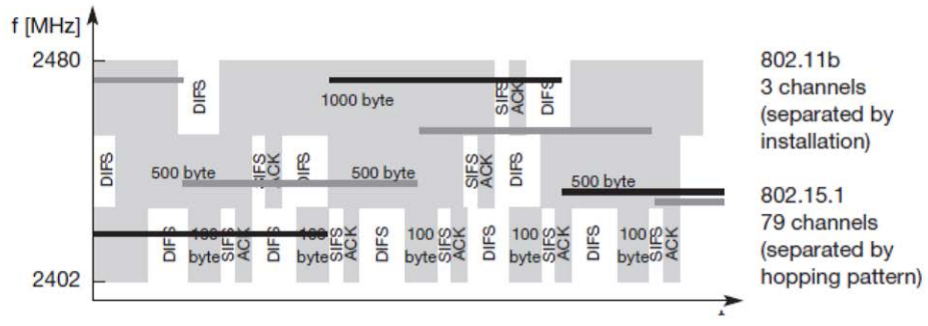


Figure 7.56
Possible interference between 802.15.1 (Bluetooth) and 802.11b

coexistence mechanism is non-collaborative in the sense that Bluetooth devices do not

have to interact with the WLAN. However, the WPAN devices can check for the occupied channels and exclude them from their list of channels used for hopping. This mechanism avoids hopping into a

channel occupied by 802.11b, but still offers enough channels for FHSS.The lower number of FHSS channels increases the interference among the WPANs due to a higher probability of collisions. However, if not too many piconets overlap this effect will be negligible. This type of interference in the crowded 2.4 GHz band is a strong argument for 5 GHz WLANs.

● **IEEE 802.15.3:** A **high-rate** study group looks for a standard providing data rates of 20 Mbit/s or greater while still working with low-power at low-cost. The standard should support isochronous data delivery, ad-hoc peer-to-peer networking, security features, and should meet the demanding requirements of portable consumer imaging and multi-media applications.

● **IEEE 802.15.4:** The fourth working group goes in the opposite direction for data rates. This group standardizes **low-rate wireless personal area networks (LR-WPAN)**, which are explained in the following section in more detail. TheZigBee consortium tries to standardize the higher layers of 802.15.4 similar to the activities of the Bluetooth consortium for 802.15.1 (ZigBee, 2002).

17.Expalin in detail about IEEE 802.15.4 – Low-rate WPANs. (L-1,CO-1)

The reason for having low data rates is the focus of the working group on extremely low power consumption enabling multi-year battery life (Callaway, 2002). Compared to 802.11 or Bluetooth, the new system should have a much lower complexity making it suitable for low-cost wireless communication (remember that Bluetooth started with similar goals with respect to the idea of cable replacement). Example **applications** include industrial control and monitoring,

smart badges, interconnection of environmental sensors, interconnection of peripherals (also an envisaged application area for Bluetooth!), remote controls etc. The new standard should offer data rates between 20 and 250 kbit/s as maximum and latencies

down to 15 ms. This is enough for many home automation and consumer electronics applications.

IEEE 802.15.4 offers two different PHY options using DSSS. The **868/915 MHz PHY** operates in Europe at 868.0–868.6 MHz and in the US at 902–928 MHz. At 868 MHz one channel is available offering a data rate of 20 kbit/s. At 915 MHz 10 channels with 40 kbit/s per channel are available (in Europe GSM uses these frequencies). The advantages of the lower frequencies are better propagation conditions. However, there is also interference in these bands as many analog

transmission systems use them. The **2.4 GHz PHY** operates at 2.4–2.4835 GHz and offers 16 channels with 250 kbit/s per channel. This PHY offers worldwide operation but suffers from interference in the 2.4 GHz ISM band and higher propagation loss. Typical devices with 1 mW output power are expected to cover a 10–20 m range. All PHY PDUs start with a 32 bit preamble for synchronization. After a start-of-packet delimiter, the PHY header indicates the length of the payload (maximum 127 bytes).

Compared to Bluetooth the **MAC layer** of 802.15.4 is much simpler. For example, no synchronous voice links are supported. MAC frames start with a 2-byte frame control field, which specifies how the rest of the frame looks and what it contains. The following 1-byte sequence number is needed to match acknowledgements with a previous data transmission. The variable address field (0–20 bytes) may contain source and/or destination addresses in various formats.

The payload is variable in length; however, the whole MAC frame may not exceed 127 bytes in length. A 16-bit FCS protects the frame. Four different MAC frames have been defined: beacon, data, acknowledgement, and MAC command. Optionally, this LR-WPAN offers a **superframe mode**. In this mode, a PAN coordinator transmits beacons in predetermined intervals (15 ms–245 s). With the help of beacons, the medium access scheme can have a period when contention

is possible and a period which is contention free. Furthermore, with beacons a slotted **CSMA/CA** is available. Without beacons standard CSMA/CA is used for medium access. Acknowledgement frames confirming a previous transmission do not use the CSMA mechanism. These frames are sent immediately following the previous packet.

IEEE 802.15.4 specifies three levels of **security**: no security, access control lists, and symmetric encryption using AES-128. Key distribution is not specified further. Security is a must for home automation or industry control applications. Up to now, the success of this standard is unclear as it is squeezed between Bluetooth, which also aims at cable replacement, and enhanced RFIDs/RF controllers.

UNIT II

PART A

1.What do you meant by topologically correct address ? (L-1,CO-2)

A router would otherwise have to store the addresses of all computers in the internet, which is obviously not feasible. As long as the receiver can be reached within its physical subnet, it gets the packets; as soon as it moves outside the subnet, a packet will not reach it. A host needs a so-called topologically correct address**.**

2.Give the significance of socket pair. (L-3,CO-2)

A TCP connection is identified by the tuple (source IP address, source port, destination IP address, destination port), also known as a **socket pair** (a socket consists of address and port). Therefore, a TCP connection cannot survive any address change. Breaking TCP connections is not an option, using even simple programs like telnet would be impossible. The mobile node would also have to notify all communication partners about the new address.

3.Give short notes about Mobile Node. **(L-1,CO-2)**

A mobile node is an end-system or router that can change its point of attachment to the internet using mobile IP. The MN keeps its IP address and can continuously communicate with any other system in the internet as long as link-layer connectivity is given. Mobile nodes are not necessarily small devices such as laptops with antennas or mobile phones; a router onboard an aircraft can be a powerful mobile node.

4.Explain in your point of view about care of address. **(L-1,CO-2)**

The COA defines the current location of the MN from an IP point of view. All IP packets sent to the MN are delivered to the COA, not directly to the IP address of the MN. Packet delivery toward the MN is done using a tunnel, as explained later. To be more precise, the COA marks the tunnel endpoint, i.e., the address where packets exit the tunnel.

5. How the Host agent acts as manager in the routing? (H-1,CO-2)

If changing the router's software is not possible, the HA could also be implemented on an arbitrary node in the subnet. One disadvantage of this solution is the double crossing of the router by the packet if the MN is in a foreign network. A packet for the MN comes in via the router; the HA sends it through the tunnel which again crosses the Finally, a home network is not necessary at all. The HA could be again on the 'router' but this time only acting as a manager for MNs belonging to a virtual home network. All MNs are always in a foreign network with this solution.

6. **How tunneling helps to deliver the packets in networking? (H-2,CO-2)**

A **tunnel** establishes a virtual pipe for data packets between a tunnel entry and a tunnel endpoint. Packets entering a tunnel are forwarded inside the tunnel and leave the tunnel unchanged. Tunneling, i.e., sending a packet through a tunnel, is achieved by using encapsulation.

7. **Give the Difference between Encapsulation& decapsulation**. **(L-1,CO-2)**

Encapsulation is the mechanism of taking a packet consisting of packet header and data and putting it into the data part of a new packet. The reverse operation, taking a packet out of the data part of another packet, is called **decapsulation**. Encapsulation and decapsulation are the operations typically performed when a packet is transferred from a higher protocol layer to a lower layer or from a lower to a higher layer respectively. Here these functions are used within the same layer.

8**. How the offset respresenting the ruting techniques? (L-2,CO-2)**

The **offset** represents the offset in bytes for the first source **routing** entry. The routing field, if present, has a variable length and contains fields for source routing.

9**.How the recursion control field works? (L-3,CO-2)**

The **recursion control** field (rec.) is an important field that additionally distinguishes GRE from IP-in-IP and minimal encapsulation. This field represents a counter that shows the number of allowed recursive encapsulations.

10. How The Triangular Protocol Works? (H-1,CO-2)

The triangle is made of the three segments, CN to HA, HA to COA/MN, and MN back to CN. With the basic mobile IP protocol all packets to the MN have to go through the HA. This can cause unnecessary overheads for the network between CN and HA, but also between HA and COA, depending on the current location of theMN.

11. Draw The Example For Weak Link And Strong Link Adhoc Network. **(L-1,CO-2)**

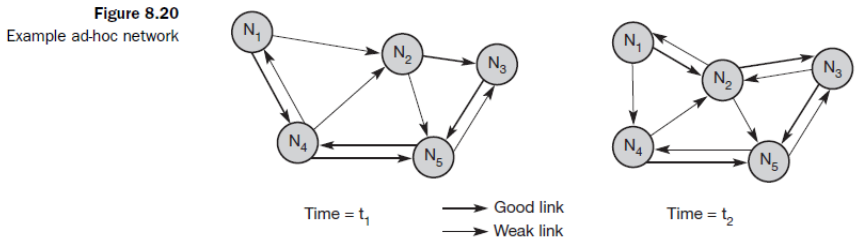


Figure 8.20
Example ad-hoc network

12.What Do You Meant By Destination sequence distance vector? (L-1,CO-2)

Destination sequence distance vector (DSDV) routing is an enhancement to distance vector routing for ad-hoc networks (Perkins, 1994). DSDV can be considered historically, however, an on-demand version (ad-hoc on-demand distance vector, AODV) is among the protocols

13.Give The Disadvantage Of Proactive Schemes. (L-1,CO-2)

A big **disadvantage** of proactive schemes are their overheads in lightly loaded networks. Independent of any real communication the algorithm continuously updates the routing tables. This generates a lot of unnecessary traffic and drains the batteries of mobile devices.

PART B

1.Explain in detail about Mobile IP. (L-1,CO-2)

The following gives an overall view of Mobile IP, and the extensions needed for the internet to support the mobility of hosts. A good reference for the original standard (RFC 2002, Perkins, 1996a) is Perkins (1997) and Solomon (1998) which describe the development of mobile IP, all packet formats, mechanisms, discussions of the protocol and alternatives etc. in detail. The new version of Mobile IP does not involve major changes in the basic architecture but corrects some minor problems (RFC 3344, Perkins, 2002). The following material requires some familiarity with Internet protocols, especially IP. A very good overview which includes detailed descriptions of classical Internet protocols is given in Stevens (1994). Many new approaches related to Internet protocols, applications, and architectures can be found in Kurose (2003).

Goals, assumptions and requirements

As shown , mobile computing is clearly the paradigm of the future. The internet is the network for global data communication with hundreds of millions of users. So why not simply use a mobile computer in the internet? The reason is quite simple: you will not receive a single packet as soon as you leave your home network, i.e., the network your computer is configured for, and reconnect your computer (wireless or wired) at another place (if no additional mechanisms are available). The reason for this is quite simple if you consider routing mechanisms on the internet. A host sends an IP packet with the header containing a destination address with other fields. The destination address not only determines the receiver of the packet, but also the physical subnet of the receiver. For example, the destination address 129.13.42.99 shows that the receiver must be connected to the physical subnet with the network prefix 129.13.42 (unless CIDR is used, RFC 1519, Fuller, 1993). Routers in the internet now look at the destination addresses of incoming packets and forward them according to internal look-up tables. To avoid an explosion of routing tables, only prefixes are stored and further optimizations are applied. A router would otherwise have to store the addresses of all computers in the internet,

which is obviously not feasible. As long as the receiver can be reached within its physical subnet, it gets the packets; as soon as it moves outside the subnet, a packet will not reach it. A host needs a so-called **topologically correct address.**

Quick 'solutions'

One might think that a quick solution to this problem would be to assign to the computer a new, topologically correct IP address. This is what many users do with the help of DHCP (see section 8.2). So moving to a new location would mean assigning a new IP address. The problem is that nobody knows about this new address. It is almost impossible to find a (mobile) host on the internet which has just changed its address.

One could argue that with the help of dynamic DNS (DDNS, RFC 2136, Vixie, 1997) an update of the mapping logical name – IP address is possible. This is what many computer users do if they have a dynamic IP address and still want to be permanently reachable using the same logical computer name. It is important to note that these considerations, indeed most of mobile IP's motivation, are important if a user wants to offer services from a mobile node, i.e., the node should act as server. Typically, the IP

address is of no special interest for service usage: in this case DHCP is sufficient. Another motivation for permanent IP addresses is emergency communication with permanent and quick reachability via the same IP address. So what about dynamically adapting the IP address with regard to the current location? The problem is that the domain name system (DNS) needs some time before it updates the internal tables necessary to map a logical name to an IP address. This approach does not work if the mobile node moves quite often. The internet and DNS have not been built for frequent updates. Just imagine millions of nodes moving at the same time. DNS could never present a consistent view of names and addresses, as it uses caching to improve scalability. It is simply too expensive to update quickly.

There is a severe problem with higher layer protocols like TCP which rely on IP addresses. Changing the IP address while still having a TCP connection open means breaking the connection. A TCP connection is identified by the tuple (source IP address, source port, destination IP address, destination port), also known as a **socket pair** (a socket consists of address and port). Therefore, a TCP connection cannot survive any address change. Breaking TCP connections is not an option, using even simple programs like telnet would be impossible. The mobile node would also have to notify all communication partners about the new address.

Another approach is the creation of specific routes to the mobile node. Routers always choose the best-fitting prefix for the routing decision. If a router now has an entry for a prefix 129.13.42 and an address 129.13.42.99, it would choose the port associated with the latter for forwarding, if a packet with the destination address 129.13.42.99 comes in. While it is theoretically possible to

change routing tables all over the world to create specific routes to a mobile node, this does not scale at all with the number of nodes in the internet. Routers are built for extremely fast forwarding, but not for fast updates of routing tables. While the first is done with special hardware support, the latter is typically a piece of software which cannot handle the burden of frequent updates. Routers are the ‘brains’ of the internet, holding the whole net together. No service provider or system administrator would allow changes to the routing tables, probably sacrificing stability, just to provide mobility for individual users.

2. Explain the Requirements of Mobile IP. (L-3,CO-2)

Since the quick 'solutions' obviously did not work, a more general architecture had to be designed. Many field trials and proprietary systems finally led to mobile IP as a standard to enable mobility in the internet. Several requirements accompanied the development of the standard:

● **Compatibility:** The installed base of Internet computers, i.e., computers running TCP/IP and connected to the internet, is huge. A new standard cannot introduce changes for applications or network protocols already in use. People still want to use their favorite browser for www and do not want to change applications just for mobility, the same holds for operating systems. Mobile IP has to be integrated into existing operating systems or at least work with them (today it is available for many platforms). Routers within the internet should not necessarily require other software. While it is possible to enhance the capabilities of some routers to support mobility,it is almost impossible to change all of them. Mobile IP has to remain compatible with all lower layers used for the standard, non-mobile, IP. Mobile IP must not require special media or MAC/LLC protocols, so it must use the same interfaces and mechanisms to access the lower layers as IP does. Finally, end-systems enhanced with a mobile IP implementation should still be able to communicate with fixed systems without mobile IP. Mobile IP has to ensure that users can still access all the other servers and systems in the internet. But that implies using the same address format and routing mechanisms.

● **Transparency:** Mobility should remain 'invisible' for many higher layer protocols and applications. Besides maybe noticing a lower bandwidth and some interruption in service, higher layers should continue to work even if the mobile computer has changed its point of attachment to the network.

For TCP this means that the computer must keep its IP address as explained above. If the interruption of the connectivity does not take too long, TCP connections survive the change of the attachment point. Problems related to the performance of TCP are discussed. Clearly, many of today's applications have not been designed for use in mobile environments, so the only effects of mobility should be a higher delay and lower bandwidth. However, there are some applications for which it is better to be 'mobility

aware'. Examples are cost-based routing or video compression. Knowing that it is currently possible to use different networks, the software could choose the cheapest one. Or if a video application knows that only a low bandwidth connection is currently available, it could use a different compression scheme. Additional mechanisms are necessary to inform these applications about mobility (Brewer, 1998).

● **Scalability and efficiency:** Introducing a new mechanism to the internet must not jeopardize its efficiency. Enhancing IP for mobility must not generate too many new messages flooding the whole network. Special care has to be taken considering the lower bandwidth of wireless links. Many mobile systems will have a wireless link to an attachment point, so only some additional packets should be necessary between a mobile system and a node in the network. Looking at the number of computers connected to the internet and at the growth rates of mobile communication, it is clear that myriad devices will participate in the internet as mobile components. Just think of cars, trucks, mobile phones, every seat in every plane around the world etc. – many of them will have some IP implementation inside and move between different networks and require mobile IP. It is crucial for a mobile IP to be scalable over a large number of participants in the whole internet, worldwide.

● **Security:** Mobility poses many security problems. The minimum requirement is that of all the messages related to the management of Mobile IP are authenticated. The IP layer must be sure that if it forwards a packet to a mobile host that this host receives the packet. The IP layer can only guarantee that the IP address of the receiver is correct. There are no ways of preventing fake IP addresses or other attacks. According to Internet philosophy, this is left to higher layers (keep the core of the internet simple, push more complex services to the edge).

The goal of a mobile IP can be summarized as: 'supporting end-system mobility while maintaining scalability, efficiency, and compatibility in all respects with existing applications and Internet protocols'.

3. How the Entities and terminology of mobile networks? (L-1,CO-2)

The following defines several entities and terms needed to understand mobile IP as defined in RFC 3344 (Perkins, 2002; was: RFC 2002, Perkins, 1996a). Figure 8.1 illustrates an example scenario.

● **Mobile node (MN):** A mobile node is an end-system or router that can change its point of attachment to the internet using mobile IP. The MN keeps its IP address and can continuously communicate with any other system in the internet as long as link-layer connectivity is given. Mobile nodes are not necessarily small devices such as laptops with antennas or mobile phones; a router onboard an aircraft can be a powerful mobile node.

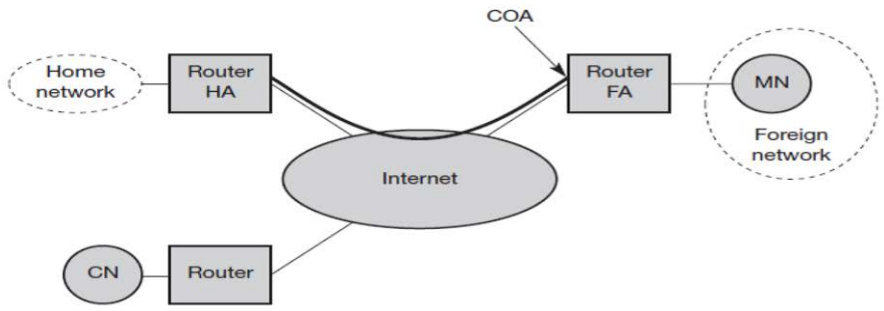


Figure 8.1
Mobile IP example network

Correspondent node (CN): At least one partner is needed for communication.

In the following the CN represents this partner for the MN. The CN can be a fixed or mobile node.

● **Home network:** The home network is the subnet the MN belongs to with respect to its IP address. No mobile IP support is needed within the home network.

● **Foreign network:** The foreign network is the current subnet the MN visits and which is not the home network.

● **Foreign agent (FA):** The FA can provide several services to the MN during its visit to the foreign network. The FA can have the COA (defined below), acting as tunnel endpoint and forwarding packets to the MN. The FA can be the default router for the MN. FAs can also provide security services because they belong to the foreign network as opposed to the MN which is only visiting.

For mobile IP functioning, FAs are not necessarily needed. Typically, an FA is implemented on a router for the subnet the MN attaches to.

● **Care-of address (COA):** The COA defines the current location of the MN from an IP point of view. All IP packets sent to the MN are delivered to the COA, not directly to the IP address of the MN. Packet delivery toward the MN is done using a tunnel, as explained later. To be more precise, the COA marks the tunnel endpoint, i.e., the address where packets exit the tunnel.

There are two different possibilities for the location of the COA:

● **Foreign agent COA:** The COA could be located at the FA, i.e., the COA is an IP address of the FA. The FA is the tunnel end-point and forwards packets to the MN. Many MN using the FA can share this COA as common COA.

● **Co-located COA:** The COA is co-located if the MN temporarily acquired an additional IP address which acts as COA. This address is now topologically correct, and the tunnel endpoint is at the MN. Co-located addresses can be acquired using services such as DHCP (see section 8.2). One problem associated with this approach is the need for additional addresses if MNs request a COA. This is not always a good idea considering the scarcity of IPv4 addresses.

● **Home agent (HA):** The HA provides several services for the MN and is located in the home network. The tunnel for packets toward the MN starts at the HA. The HA maintains a location registry, i.e., it is informed of the MN's location by the current COA. Three alternatives for the implementation of an HA exist.

● The HA can be implemented on a router that is responsible for the home network. This is obviously the best position, because without optimizations to mobile IP, all packets for the MN have to go through the router anyway.

● If changing the router's software is not possible, the HA could also be implemented on an arbitrary node in the subnet. One disadvantage of this solution is the double crossing of the router by the packet if the MN is in a foreign network. A packet for the MN comes in via the router; the HA sends it through the tunnel which again crosses the Finally, a home network is not necessary at all. The HA could be again on the 'router' but this time only acting as a manager for MNs belonging to a virtual home network. All MNs are always in a foreign network with this solution.

The example network in Figure 8.1 shows the following situation: A CN is connected via a router to the internet, as are the home network and the foreign network. The HA is

implemented on the router connecting the home network with the internet, an FA is implemented on the router to the foreign network. The MN is currently in the foreign network. The tunnel for packets toward the
MN starts at the HA and ends at the FA, for the FA has the COA in this example.

4. How the IP packet delivery is achieved in WLAN? (H-1,CO-2)

Figure 8.2 illustrates packet delivery to and from the MN using the example network of Figure 8.1. A correspondent node CN wants to send an IP packet to the MN. One of the requirements of mobile IP was to support hiding the mobility of the MN. CN does not need to know anything about the MN's current location and sends the packet as usual to the IP address of MN (step 1). This means that CN sends an IP packet with MN as a destination address and CN as a source address. The internet, not having information on the current location of MN, routes the packet to the router responsible for the home network of MN. This is
done using the standard routing mechanisms of the internet. The HA now intercepts the packet, knowing that MN is currently not in its home network. The packet is not forwarded into the subnet as usual, but encapsulated and tunnelled to the COA. A new header is put in front of the old IP header showing the COA as new destination and HA as source of the encapsulated packet (step 2). (Tunneling and encapsulation is described in more detail in section 8.1.6.) The foreign agent now decapsulates the packet, i.e., removes the additional header, and forwards the original packet with CN as source and MN as destination to the MN (step 3). Again, for the MN mobility is not visible. It receives the packet with the same sender and receiver address as it would have
done in the home network.

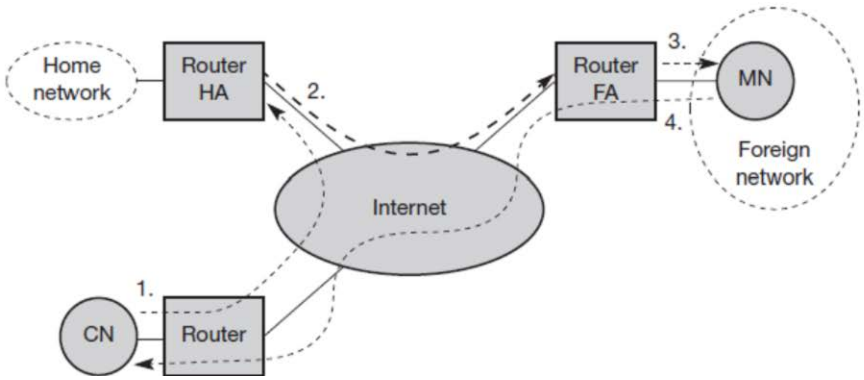


Figure 8.2
Packet delivery to and from the mobile node

At first glance, sending packets from the MN to the CN is much simpler; problems are discussed in section. The MN sends the packet as usual with its own fixed IP address as source and CN's address as destination (step 4). The router with the FA acts as default router and forwards the packet in the same way as it would do for any other node in the foreign network. As long as CN is a fixed node the remainder is in the fixed internet as usual. If CN were also a mobile node residing in a foreign network, the same mechanisms as described in steps 1 through 3 would apply now in the other direction.

The following sections present some additional mechanisms needed for mobile IP to work, some enhancements to the protocol, and some efficiency and security problems.

Agent discovery

One initial problem of an MN after moving is how to find a foreign agent. How does the MN discover that it has moved? For this purpose mobile IP describes two methods: agent advertisement and agent solicitation, which are in fact router discovery methods plus extensions.

Agent advertisement

For the first method, foreign agents and home agents advertise their presence periodically using special **agent advertisement** messages. These advertisement messages can be seen as a beacon broadcast into the subnet. For these advertisements Internet control message protocol (ICMP) messages according to RFC 1256 (Deering, 1991) are used with some mobility extensions. Routers in the fixed network implementing this standard also advertise their routing service periodically to the attached links.

The agent advertisement packet according to RFC 1256 with the extension for mobility is shown in Figure . The upper part represents the ICMP packet while the lower part is

the extension needed for mobility. The fields necessary on lower layers for the agent advertisement are not shown in this figure. Clearly, mobile nodes must be reached with the appropriate link layer address. The TTL field of the IP packet is set to 1 for all advertisements to avoid forwarding them. The IP destination address according to standard router advertisements can be either set to 224.0.0.1, which is the multicast address for all systems on a link (Deering, 1989), or to the broadcast address 255.255.255.255. The fields in the ICMP part are defined as follows. The **type** is set to 9, the **code** can be 0, if the agent also routes traffic from non-mobile nodes, or 16, if it does not route anything other than mobile traffic. Foreign agents are at least required to forward packets from the mobile node. The number of addresses advertised with this packet is in **#addresses** while the **addresses** themselves follow as shown. **Lifetime** denotes the length of time this advertisement is valid. **Preference** levels for each address help a node to choose the router that is the most eager one to get a new node.

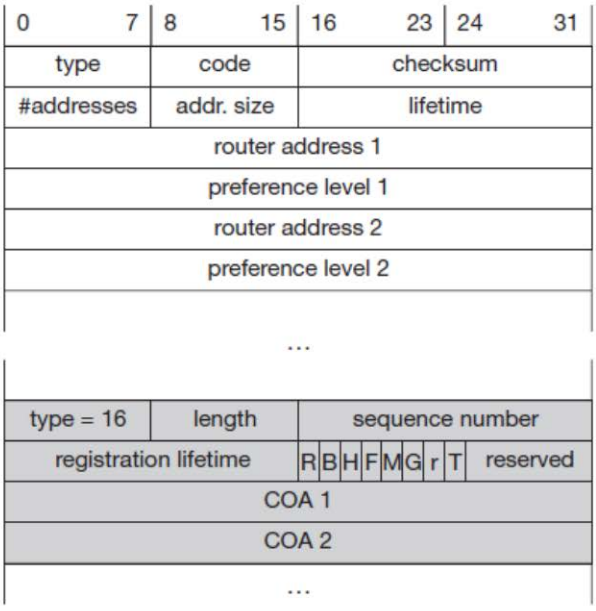


Figure 8.3
Agent advertisement packet (RFC 1256 + mobility extension)

The difference compared with standard ICMP advertisements is what happens after the router addresses. This extension for mobility has the following fields defined: **type** is set to 16, **length** depends on the number of COAs provided with the message and equals 6 + 4*(number of addresses). An agent shows the total number of advertisements sent

since initialization in the **sequence number**. By the **registration lifetime** the agent can specify the maximum lifetime

in seconds a node can request during registration as explained in section 8.1.5. The following bits specify the characteristics of an agent in detail. The **R** bit (registration) shows, if a registration with this agent is required even when using a colocated COA at the MN. If the agent is currently too busy to accept new registrations it can set the **B** bit. The following two bits denote if the agent offers services as a home agent (**H**) or foreign agent (**F**) on the link where the advertisement has been sent. Bits M and G specify the method of encapsulation used for the tunnel as explained in section 8.1.6. While IP-in-IP encapsulation is the mandatory standard, **M** can specify minimal encapsulation and **G** generic routing encapsulation. In the first version of mobile IP (RFC 2002) the **V** bit specified the use of header compression according to RFC 1144 (Jacobson, 1990). Now the field **r** at the same bit position is set to zero and must be ignored. The new field **T** indicates that reverse tunneling (see section 8.1.8) is supported by the FA. The following fields contain the **COAs** advertised. A foreign agent setting the F bit must advertise at least one COA. Further details and special extensions can be found in Perkins (1997) and RFC 3220. A mobile node in a subnet can now receive agent advertisements from either its home agent or a foreign agent. This is one way for the MN to discover its location.

Agent solicitation

If no agent advertisements are present or the inter-arrival time is too high, and an MN has not received a COA by other means, e.g., DHCP as discussed in section 8.2, the mobile node must send **agent solicitations**. These solicitations are again based on RFC 1256 for router solicitations. Care must be taken to ensure that these solicitation messages do not flood the network, but basically an MN can search for an FA endlessly sending out solicitation messages. Typically, a mobile node can send out three solicitations, one per second, as soon as it enters a new network. It should be noted that in highly dynamic wireless networks with moving MNs and probably with applications requiring continuous packet streams even one second intervals between solicitation messages might be too long. Before an MN even gets a new address many packets will be lost without additional mechanisms.

If a node does not receive an answer to its solicitations it must decrease the rate of solicitations exponentially to avoid flooding the network until it reaches a maximum interval between solicitations (typically one minute). Discovering a new agent can be done anytime, not just if the MN is not connected to one. Consider the case that an MN is looking for a better connection while still sending via the old path. This is the case while moving through several cells of different wireless networks. After these steps of advertisements or solicitations the MN can now receive a COA, either one for an FA or a co-located COA. The MN knows its location (home network or foreign network) and the capabilities of the agent (if needed). The next step for the MN is the registration with the HA if the MN is in a foreign network as described in the following.

Registration

Having received a COA, the MN has to register with the HA. The main purpose of the registration is to inform the HA of the current location for correct forwarding of packets. Registration can be done in two different ways depending on the location of the COA.

● If the COA is at the FA, registration is done as illustrated in Figure 8.4 (left). The MN sends its registration request containing the COA (see Figure 8.5) to the FA which is forwarding the request to the HA. The HA now sets up a **mobility binding** containing the mobile node's home IP address and the current COA. Additionally, the mobility binding contains the lifetime of the registration which is negotiated during the registration process. Registration expires automatically after the lifetime and is deleted; so, an MN should reregister before expiration. This mechanism is necessary to avoid mobility bindings which are no longer used. After setting up the mobility binding, the HA sends a reply message back to the FA which forwards it to the MN.

● If the COA is co-located, registration can be simpler, as shown in Figure 8.4 (right). The MN may send the request directly to the HA and vice versa. This, by the way, is also the registration procedure for MNs returning to their home network. Here they also register directly with the HA. However, if the MN received an agent advertisement from the FA it should register via

this FA if the R bit is set in the advertisement.

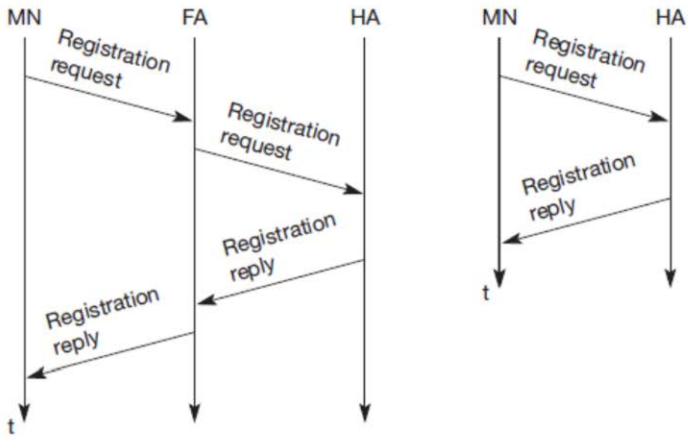


Figure 8.4 Registration of a mobile node via the FA or directly with the HA

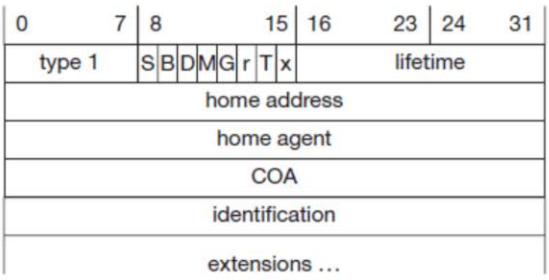
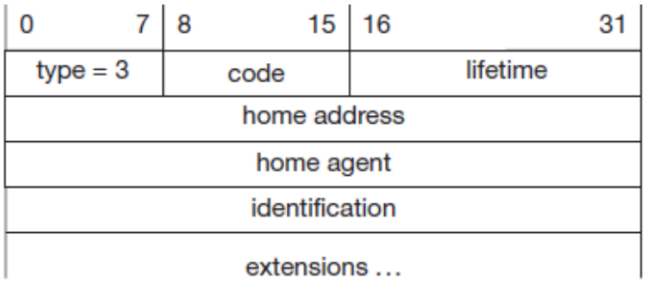


Figure 8.5 Registration request

UDP packets are used for **registration requests**. The IP source address of the packet is set to the interface address of the MN, the IP destination address is that of the FA or HA (depending on the location of the COA). The UDP destination port is set to 434. UDP is used because of low overheads and better performance compared to TCP in wireless environments. The fields relevant for mobile IP registration requests follow as UDP data (see Figure 8.6). The fields are defined as follows.

The first field **type** is set to 1 for a registration request. With the **S** bit an MN can specify if it wants the HA to retain prior mobility bindings. This allows for simultaneous bindings. The following bits denote the requested behavior for packet forwarding. Setting the **B** bit generally indicates that an MN also wants to receive the broadcast packets which have been received by the HA in the home network. A more detailed description of how to filter broadcast messages which are not needed by the MN can be found in Perkins (1997). If an MN uses a co-located COA, it also takes care of the decapsulation at the tunnel endpoint. The **D** bit indicates this behavior. As already defined for agent advertisements, the following bits **M** and **G** denote the use of minimal encapsulation or generic routing encapsulation, respectively. **T** indicates reverse tunneling, **r** and **x** are set to zero.

Figure 8.6
Registration reply



Lifetime denotes the validity of the registration in seconds. A value of zero indicates deregistration; all bits set indicates infinity. The **home address** is the fixed IP address of the MN, **home agent** is the IP address of the HA, and **COA** represents the tunnel endpoint. The 64 bit **identification** is generated by the MN to identify a request and match it with registration replies. This field is used for protection against replay attacks of registrations. The **extensions** must at least contain parameters for authentication.

A **registration reply**, which is conveyed in a UDP packet, contains a **type** field set to 3 and a **code** indicating the result of the registration request. Table 8.1 gives some example codes.

Table 8.1 Example registration reply codes

Registration	Code	Explanation
successful	0	registration accepted
	1	registration accepted, but simultaneous mobility bindings unsupported
denied by FA	65	administratively prohibited
	66	insufficient resources
	67	mobile node failed authentication
	68	home agent failed authentication
	69	requested lifetime too long
denied by HA	129	administratively prohibited
	130	insufficient resources
	131	mobile node failed authentication
	132	foreign agent failed authentication
	133	registration identification mismatch
	135	too many simultaneous mobility bindings

The **lifetime** field indicates how many seconds the registration is valid if it was successful. **Home address** and **home agent** are the addresses of the MN and the HA, respectively. The 64-bit **identification** is used to match registration requests with replies. The value is based on the identification field from the registration and the authentication method. Again, the **extensions** must at least contain parameters for authentication.

5.. Explain about Tunneling and encapsulation techniques. (L-1,CO-2)

The following describes the mechanisms used for forwarding packets between the HA and the COA, as shown in Figure 8.2, step 2. A **tunnel** establishes a virtual pipe for data packets between a tunnel entry and a tunnel endpoint. Packets entering a tunnel are forwarded inside the tunnel and leave the tunnel unchanged. Tunneling, i.e., sending a packet through a tunnel, is achieved by

using encapsulation. **Encapsulation** is the mechanism of taking a packet consisting of packet header and data and putting it into the data part of a new packet. The reverse operation, taking a packet out of the data part of another packet, is called **decapsulation**. Encapsulation and decapsulation are the operations typically performed when a packet is transferred from a higher protocol layer to a lower layer or from a lower to a higher layer respectively. Here these functions are used within the same layer. This mechanism is shown in Figure 8.7 and describes exactly what the HA at the tunnel entry does. The HA takes the original packet with the MN as destination, puts it into the data part of a new packet and sets the new IP header in such a way that the packet is routed to the COA. The new header is also called the **outer header** for obvious reasons. Additionally, there is an **inner header** which can be identical to the original header as this is the case for IP-in-IP encapsulation, or the inner header can be computed during encapsulation.

IP-in-IP encapsulation

There are different ways of performing the encapsulation needed for the tunnel between HA and COA. Mandatory for mobile IP is **IP-in-IP encapsulation** as specified in RFC 2003 (Perkins, 1996b). Figure 8.8 shows a packet inside the tunnel. The fields follow the standard specification of the IP protocol as defined

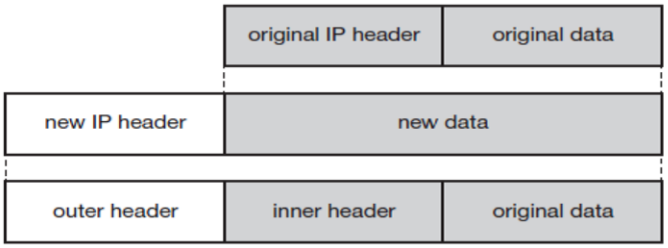


Figure 8.7
IP encapsulation

Figure 8.8
IP-in-IP encapsulation

ver.	IHL	DS (TOS)	length
IP identification		flags	fragment offset
TTL	*IP-in-IP*	IP checksum	
IP address of HA			
Care-of address of COA			
ver.	IHL	DS (TOS)	length
IP identification		flags	fragment offset
TTL	lay. 4 prot.	IP checksum	
IP address of CN			
IP address of MN			
TCP/UDP/ … payload			

in RFC 791 (Postel, 1981) and the new interpretation of the former TOS, now DS field in the context of differentiated services (RFC 2474, Nichols, 1998). The fields of the outer header are set as follows. The version field **ver** is 4 for IP version 4, the internet header length (**IHL**) denotes the length of the outer header in 32 bit words. **DS(TOS)** is just copied from the inner header, the **length** field covers the complete encapsulated packet. The fields up to TTL have no special meaning for mobile IP and are set according to RFC 791. **TTL** must be high enough so the packet can reach the tunnel endpoint. The next field, here

denoted with **IP-in-IP**, is the type of the protocol used in the IP payload. This field is set to 4, the protocol type for IPv4 because again an IPv4 packet follows after this outer header. IP **checksum** is calculated as usual. The next fields are the tunnel entry as source address (the **IP address of the HA**) and the tunnel exit point as destination address (the **COA**). If no options follow the outer header, the inner header starts with

the same fields as just explained. This header remains almost unchanged during encapsulation, thus showing the original sender CN and the receiver MN of the packet. The only change is TTL which is decremented by 1. This means that the whole tunnel is considered a single hop from the original packet's point of view. This
is a very important feature of tunneling as it allows the MN to behave as if it were attached to the home network. No matter how many real hops the packet has to take in the tunnel, it is just one (logical) hop away for the MN. Finally, the payload follows the two headers.

Minimal encapsulation

As seen with IP-in-IP encapsulation, several fields are redundant. For example, TOS is just copied, fragmentation is often not needed etc. Therefore, **minimal encapsulation** (RFC 2004) as shown in Figure 8.9 is an optional encapsulation method for mobile IP (Perkins, 1996c). The tunnel entry point and endpoint are specified. In this case, the field for the type of the following header contains the

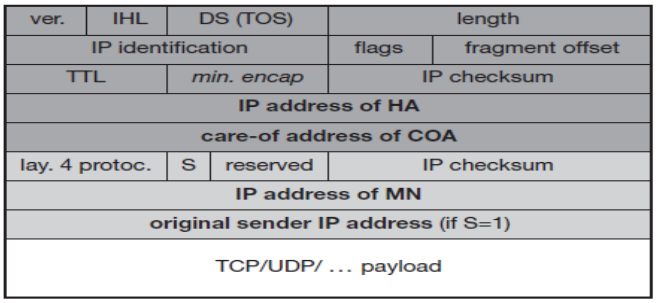


Figure 8.9
Minimal encapsulation

value 55 for the minimal encapsulation protocol. The inner header is different for minimal encapsulation. The type of the following protocol and the address of the MN are needed. If the **S** bit is set, the original sender address of the CN is included as omitting the source is quite often not an option. No field for fragmentation offset is left in the inner header and minimal encapsulation does not work with already fragmented packets.

Generic routing encapsulation

While IP-in-IP encapsulation and minimal encapsulation work only for IP, the following encapsulation scheme also supports other network layer protocols in addition to IP. **Generic routing encapsulation** (GRE) allows the encapsulation of packets of one protocol suite into the payload portion of a packet of another protocol suite (Hanks, 1994). Figure 8.10 shows this procedure. The packet of one protocol suite with the original packet header and data is taken and a new GRE header is prepended. Together this forms the new data part of the new packet. Finally, the header of the second protocol suite is put in front. Figure 8.11 shows on the left side the fields of a packet inside the tunnel between home agent and COA using GRE as an encapsulation scheme according to RFC 1701. The outer header is the standard IP header with HA as source address and COA as destination address. The protocol type used in this outer IP

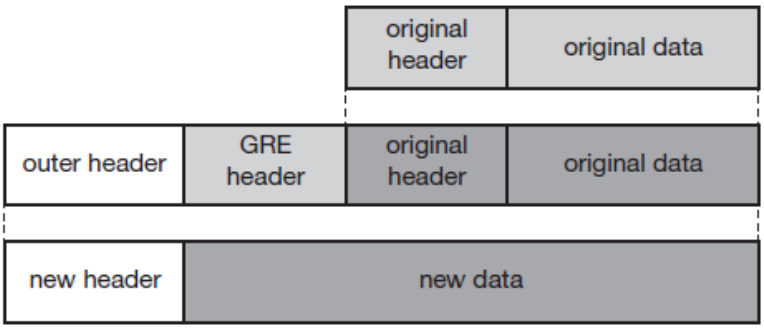
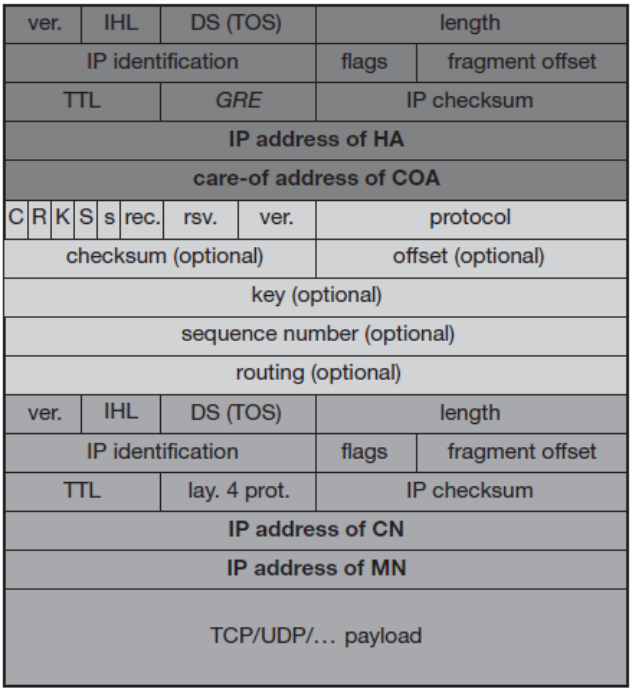


Figure 8.10
Generic routing encapsulation

Figure 8.11
Protocol fields for GRE according to RFC 1701



header is 47 for GRE. The other fields of the outer packet, such as TTL and TOS, may be copied from the original IP header. However, the TTL must be decremented by 1 when the packet is decapsulated to prevent indefinite forwarding. The GRE header starts with several flags indicating if certain fields are present or not. A minimal GRE header uses only 4 bytes; nevertheless, GRE is flexible enough to include several mechanisms in its header. The **C** bit indicates if the checksum field is present and contains valid information. If **C** is set, the **checksum** field contains a valid IP checksum of the GRE header and the payload. The **R** bit indicates if the offset and routing fields are present and contain valid information. The **offset** represents the offset in bytes for the first source **routing** entry. The routing field, if present, has a variable length and contains fields for source routing. If the C bit is set, the offset field is also present and, vice versa, if the R bit is set, the checksum field must be present. The only reason for this is to align the following fields to 4 bytes. The checksum field is valid only if C is set, and the offset field is valid only if R is set respectively. GRE also offers a **key** field which may be used for authentication. If this field is present, the **K** bit is set. However, the authentication algorithms are not further specified by GRE. The sequence number bit **S**

indicates if the **sequence** number field is present, if the s bit is set, strict source routing is used. Sequence
numbers may be used by a decapsulator to restore packet order. This can be important, if a protocol guaranteeing in-order transmission is encapsulated and

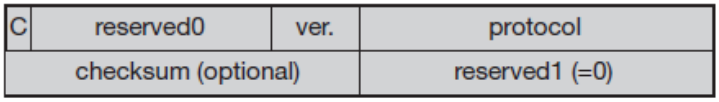


Figure 8.12
Protocol fields for GRE according to RFC 2784

transferred using a protocol which does not guarantee in-order delivery, e.g., IP. Now the decapsulator at the tunnel exit must restore the sequence to maintain the characteristic of the protocol. The **recursion control** field (rec.) is an important field that additionally distinguishes
GRE from IP-in-IP and minimal encapsulation. This field represents a counter that shows the number of allowed recursive encapsulations. As soon as a packet arrives at an encapsulator it checks whether this field equals zero. If the field is not zero, additional encapsulation is allowed – the packet is encapsulated and the field decremented by one. Otherwise the packet will most likely be discarded. This mechanism prevents indefinite recursive encasulation which
might happen with the other schemes if tunnels are set up improperly (e.g., several tunnels forming a loop). The default value of this field should be 0, thus allowing only one level of encapsulation. The following **reserved** fields must be zero and are ignored on reception. The
version field contains 0 for the GRE version. The following 2 byte **protocol** field represents the protocol of the packet following the GRE header. Several values have been defined, e.g., 0 × 6558 for transparent Ethernet bridging using a GRE tunnel. In the case of a mobile IP tunnel, the protocol field contains 0 × 800 for IP. The standard header of the original packet follows with the source address of the correspondent node and the destination address of the mobile node.
Figure 8.12 shows the simplified header of GRE following RFC 2784 (Farinacci, 2000), which is a more generalized version of GRE compared to RFC 1701. This version does

not address mutual encapsulation and ignores several protocol-specific nuances on purpose. The field **C** indicates again if a checksum is present. The next 5 bits are set to zero, then 7 reserved bits follow. The **version** field contains the value zero. The **protocol** type, again, defines the protocol of the payload following RFC 3232 (Reynolds, 2002). If the flag C is set, then **checksum** field and a field called reserved1 follows. The latter field is constant zero set to zero follow. RFC 2784 deprecates several fields of RFC 1701, but can interoperate with RFC 1701-compliant implementations.

6.. Explain the Optimizations used the mobile IP Networks. (L-2,CO-2)

Imagine the following scenario. A Japanese and a German meet at a conference on Hawaii. Both want to use their laptops for exchanging data, both run mobile IP for mobility support. Now recall Figure 8.2 and think of the way the packets between both computers take. If the Japanese sends a packet to the German, his computer sends the data to the HA of the German, i.e., from Hawaii to Germany. The HA in Germany now encapsulates the packets and tunnels them to the COA of the German laptop on Hawaii. This means that although the computers might be only meters away, the packets have to travel around the world! This inefficient behavior of a nonoptimized mobile IP is called **triangular routing**. The triangle is made of the three segments, CN to HA, HA to COA/MN, and MN back to CN. With the basic mobile IP protocol all packets to the MN have to go through the HA. This can cause unnecessary overheads for the network between CN and HA, but also between HA and COA, depending on the current location of the

MN. As the example shows, latency can increase dramatically. This is particularly unfortunate if the MNs and HAs are separated by, e.g., transatlantic links. One way to optimize the route is to inform the CN of the current location of the MN. The CN can learn the location by caching it in a **binding cache** which is a part of the local routing table for the CN. The appropriate entity to

inform the CN of the location is the HA. The optimized mobile IP protocol needs four additional messages.

● **Binding request:**

Any node that wants to know the current location of an MN can send a binding request to the HA. The HA can check if the MN has allowed dissemination of its current location. If the HA is allowed to reveal the location it sends back a binding update.

● **Binding update:**

This message sent by the HA to CNs reveals the current location of an MN. The message contains the fixed IP address of the MN and the COA. The binding update can request an acknowledgement.

● **Binding acknowledgement:**

If requested, a node returns this acknowledgement after receiving a binding update message.

● **Binding warning:**

If a node decapsulates a packet for an MN, but it is not the current FA for this MN, this node sends a binding warning. The warning contains MN's home address and a target node address, i.e., the address of the node that has tried to send the packet to this MN. The recipient of the warning then knows that the target node could benefit from obtaining a fresh
binding for the MN. The recipient can be the HA, so the HA should now send a binding update to the node that obviously has a wrong COA for the MN.

Figure 8.13 explains these additional four messages together with the case of an MN changing its FA. The CN can request the current location from the HA. If allowed by the MN, the HA returns the COA of the MN via an update message. The CN acknowledges this update message and stores the mobility binding. Now
the CN can send its data directly to the current foreign agent FAold. FAold forwards the packets to the MN. This scenario shows a COA located at an FA. Encapsulation of data for tunneling to the COA is now done by the CN, not the HA. The MN might now change

its location and register with a new foreign agent, FAnew. This registration is also forwarded to the HA to update its location database. Furthermore, FAnew informs FAold about the new registration of MN.

MN's registration message contains the address of FAold for this purpose. Passing this information is achieved via an update message, which is acknowledged by FAold. Registration replies are not shown in this scenario. Without the information provided by the new FA, the old FA would not get to know anything about the new location of MN. In this case, CN does not know anything about the new location, so it still tunnels its packets for MN to the old FA, FAold. This FA now notices packets with destination MN, but also knows that it is not the current

FA of MN. FAold might now forward these packets to the new COA of MN which is FAnew in this example. This forwarding of packets is another optimization of the basic Mobile IP providing **smooth handovers**. Without this optimization, all packets in transit would be lost while the MN moves from one FA to another. With TCP as the higher layer protocol this would result in severe performance degradation .

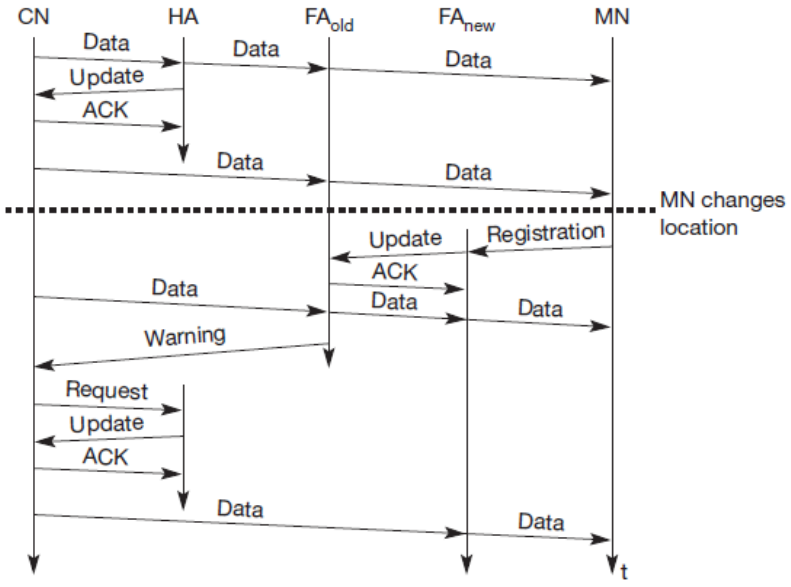


Figure 8.13 Change of the foreign agent with an optimized mobile IP

To tell CN that it has a stale binding cache, FAold sends, in this example, a binding warning essage to CN. CN then requests a binding update. (The warning could also be directly sent to the HA triggering an update). The HA sends an update to inform the CN

about the new location, which is acknowledged. Now CN can send its packets directly to FAnew, again avoiding triangular routing. Unfortunately, this optimization of mobile IP to avoid triangular routing

causes several security problems (e.g., tunnel hijacking) as discussed in Montenegro (1998). Not all users of mobile communication systems want to reveal their current 'location' (in the sense of an IP subnet) to a communication partner.

7..Explain the Reverse tunneling techniques used in WLAN.. (L-1,CO-2)

At first glance, the return path from the MN to the CN shown in Figure 8.2 looks quite simple. The MN can directly send its packets to the CN as in any other standard IP situation. The destination address in the packets is that of CN. But there are several severe problems associated with this simple solution.

● **Firewalls:**

Almost all companies and many other institutions secure their internal networks (intranet) connected to the internet with the help of a firewall. All data to and from the intranet must pass through the firewall. Besides many other functions, firewalls can be set up to filter out malicious

addresses from an administrator's point of view. Quite often firewalls only allow packets with topologically correct addresses to pass. This provides at least a first and simple protection against misconfigured systems of unknown addresses. However, MN still sends packets with its fixed IP address as source which is not topologically correct in a foreign network. Firewalls often filter packets coming from outside containing a source address from computers of the internal network. This avoids other computers that could use internal addresses and claim to be internal computers. However, this also implies that an MN cannot send a packet to a computer residing in its home network. Altogether, this means that not only does the destination address matter for forwarding IP packets, but also the source address due to security concerns. Further omplications arise through the use of private addresses inside the intranet and the translation into global addresses when communicating with the internet. This **network address**

translation (NAT, network address translator, RFC 3022, Srisuresh, 2001) is used by many companies to
hide internal resources (routers, computers, printers etc.) and to use only some globally available addresses (Levkowetz, 2002, tries to solve the problems arising when using NAT together with mobile IP).

● **Multi-cast:**

Reverse tunnels are needed for the MN to participate in a multicast group. While the nodes in the home network might participate in a multi-cast group, an MN in a foreign network cannot transmit multi-cast packets in a way that they emanate from its home network without a reverse tunnel. The foreign network might not even provide the technical infrastructure for multi-cast communication (multi-cast backbone, Mbone).

● **TTL:**

Consider an MN sending packets with a certain TTL while still in its home network. The TTL might be low enough so that no packet is transmitted outside a certain region. If the MN now moves to a foreign network, this TTL might be too low for the packets to reach the same nodes as before. Mobile IP is no longer transparent if a user has to adjust the TTL while moving. A reverse tunnel is needed that represents only one hop, no matter how many hops are really needed from the foreign to the home network. All these considerations led to RFC 2344 (Montenegro, 1998) defining reverse tunneling as an extension to mobile IP. The new RFC 3024 (Montenegro, 2001) renders RFC 2344 obsolete but comprises only some minor changes for
the original standard. The RFC was designed backwards-compatible to mobile IP and defines topologically correct reverse tunneling as necessary to handle the problems described above. Reverse tunneling was added as an option to mobile IP in the new standard (RFC 3344).

Obviously, reverse tunneling now creates a triangular routing problem in the reverse direction. All packets from an MN to a CN go through the HA. RFC 3024 does not offer a solution for this reverse triangular routing, because it is not clear if the CN can

decapsulate packets. Remember that mobile IP should work together with all traditional, non-mobile IP nodes. Therefore, one cannot assume that a CN is able to be a tunnel endpoint. Reverse tunneling also raises several security issues which have not been really solved up to now. For example, tunnels starting in the private network of a company and reaching out into the internet could be hijacked and abused for
sending packets through a firewall. It is not clear if companies would allow for setting up tunnels through a firewall without further checking of packets. It is more likely that a company will set up a special virtual network for visiting mobile nodes outside the firewall with full connectivity to the internet. This allows guests to use their mobile equipment, and at the same time, today's security standards are maintained. Initial architectures integrating mobility and security aspects within firewalls exist (Mink, 2000a and b).

8.Explain in detail about IPv6. (L-1,CO-2)

While mobile IP was originally designed for IP version 4, IP version 6 (Deering, 1998) makes life much easier. Several mechanisms that had to be specified separately for mobility support come free in IPv6 (Perkins, 1996d), (Johnson, 2002b). One issue is security with regard to authentication, which is now a required feature for all IPv6 nodes. No special mechanisms as add-ons are needed for securing mobile IP registration. Every IPv6 node masters address autoconfiguration – the mechanisms for acquiring a COA are already built in. Neighbor
discovery as a mechanism mandatory for every node is also included in the specification; special foreign agents are no longer needed to advertise services. Combining the features of autoconfiguration and neighbor discovery means that every mobile node is able to create or obtain a topologically correct address for the current point of attachment.

Every IPv6 node can send binding updates to another node, so the MN can send its current COA directly to the CN and HA. These mechanisms are an integral part of IPv6.

A soft handover is possible with IPv6. The MN sends its new COA to the old router servicing the MN at the old COA, and the old router encapsulates all incoming packets for the MN and forwards them to the new COA.

Altogether, mobile IP in IPv6 networks requires very few additional mechanisms of a CN, MN, and HA. The FA is not needed any more. A CN only has to be able to process binding updates, i.e., to create or to update an entry in the routing cache. The MN itself has to be able to decapsulate packets, to detect when it needs a new COA, and to determine when to send binding updates to the HA and CN. A HA must be able to encapsulate packets. However, IPv6 does

not solve any firewall or privacy problems. Additional mechanisms on higher layers are needed for this.

IP micro-mobility support

Mobile IP exhibits several problems regarding the duration of handover and the scalability of the registration procedure. Assuming a large number of mobile devices changing networks quite frequently, a high load on the home agents as well as on the networks is generated by registration and binding update messages. IP micro-mobility protocols can complement mobile IP by offering fast and almost seamless handover control in limited geographical areas.

Consider a client arriving with his or her laptop at the customer's premises. The home agent only has to know an entry point to the customer's network, not the details within this network. The entry point acts as the current location. Changes in the location within the customer's network should be handled locally to minimize network traffic and to speed-up local handover. The basic

underlying idea is the same for all micro-mobility protocols: Keep the frequent updates generated by local changes of the points of attachment away from the home network and only inform the home agent about major changes, i.e., changes of a region. In some sense all micro-mobility protocols establish a hierarchy. However, the debate is still going on if micro-mobility apects should really be handled on the IP layer or if layer 2 is the better place for it. Layer 2 mobility support would comprise, e.g., the inter access point protocol (IAPP) of 802.11 WLANs or the mobility support mechanisms of mobile phone systems. The following presents three of the most prominent approaches,

which should be seen neither as standards nor as final solutions of the micro-mobility problems. Campbell (2002) presents a comparison of the three approaches.

Cellular IP

Cellular IP (Valko, 1999), (Campbell, 2000) provides local handovers without renewed registration by instaling a single **cellular IP gateway (CIPGW)** for each domain, which acts to the outside world as a foreign agent (see Figure 8.14). Inside the cellular IP domain, all nodes collect routing information for accessing MNs based on the origin of packets sent by the MNs towards the CIPGW. Soft handovers are achieved by allowing simultaneous forwarding of packets destined for a mobile node along multiple paths. A mobile node moving between adjacent cells will temporarily be able to receive packets via both old and new **base stations (BS)** if this is supported by the lower protocol layers.

Concerning the manageability of cellular IP, it has to be noted that the approach has a simple and elegant architecture and is mostly self-configuring. However, mobile IP tunnels could be controlled more easily if the CIPGW was integrated into a firewall, but there are no detailed specifications in (Campbell, 2000) regarding such integration. Cellular IP requires changes to the basic mobile IP protocol and is not transparent to existing systems. The foreign network's

routing tables are changed based on messages sent by mobile nodes. These should not be trusted blindly even if they have been authenticated. This could be exploited by systems in the foreign network for wiretapping packets

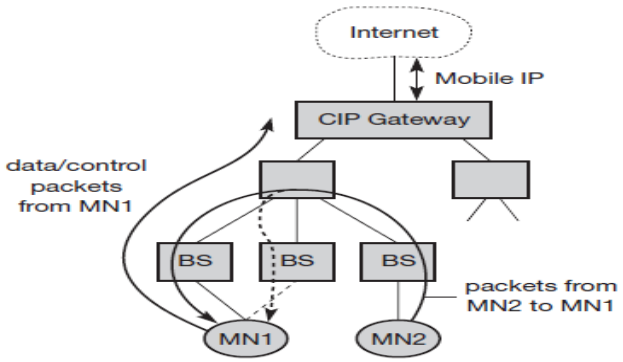


Figure 8.14
Basic architecture of cellular IP

destined for an MN by sending packets to the CIPGW with the source address set to the MN's address. In enterprise scenarios requiring basic communications security, this may not be acceptable.

Advantage

● Manageability: Cellular IP is mostly self-configuring, and integration of the CIPGW into a firewall would facilitate administration of mobility-related functionality. This is, however, not explicitly specified in (Campbell, 2000).

Disadvantages

● Efficiency: Additional network load is induced by forwarding packets on multiple paths.

● Transparency: Changes to MNs are required.

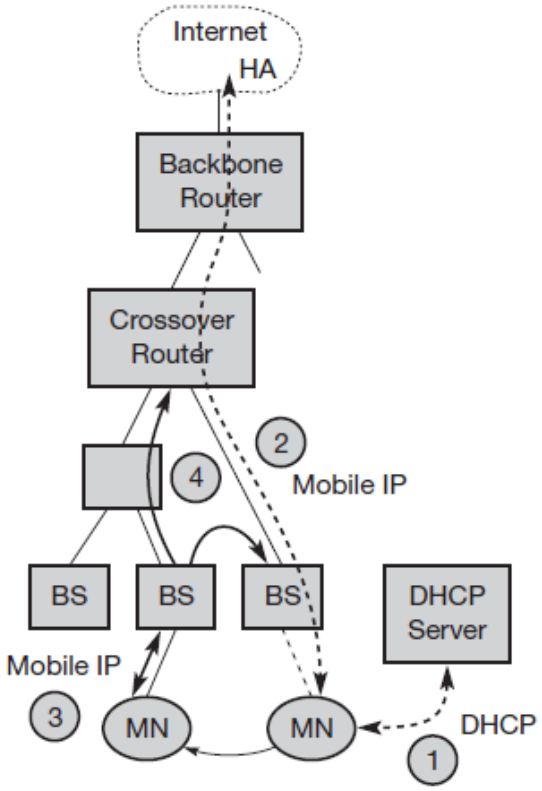
● Security: Routing tables are changed based on messages sent by mobile nodes. Additionally, all systems in the network can easily obtain a copy of all packets destined for an MN by sending packets with the MN's source address to the CIPGW.

9..Brief about Hawaii and its evolution technqiues. (L-1,CO-2)

HAWAII (Handoff-Aware Wireless Access Internet Infrastructure, Ramjee, 1999) tries to keep micro-mobility support as transparent as possible for both home agents and mobile nodes (which have to support route optimization). Its concrete goals are performance and reliability improvements and support for quality of service mechanisms. On entering an HAWAII domain, a mobile node obtains a co-located COA (see Figure 8.15, step 1) and registers with the HA

(step 2). Additionally, when moving to another cell inside the foreign domain, the MN sends a registration request to the new base station as to a foreign agent

Figure 8.15
Basic architecture of HAWAII



(step 3), thus mixing the concepts of co-located COA and foreign agent COA. The base station intercepts the registration request and sends out a handoff update message, which reconfigures all routers on the paths from the old and new base station to the so-called crossover router (step 4). When routing has been reconfigured successfully, the base station sends a registration reply to the mobile node, again as if it were a foreign agent.

The use of challenge-response extensions for authenticating a mobile node is mandatory. In contrast to cellular IP, routing changes are always initiated by the foreign domain's nfrastructure, and the corresponding messages could be authenticated, e.g., by means of an IPSec authentication header (AH; RFC 2402, Kent, 1998), reducing the risk of malicious rerouting of traffic initiated by bogus mobile hosts. However, this is not explicitly specified in Ramjee 1999). HAWAII claims to be mostly transparent to mobile nodes, but this claim has to be regarded with some caution as the requirement to

support a co-located care-ofaddress as well as to interact with foreign agents could cause difficulties with some mobile nodes.

Advantages

● Security: Challenge-response extensions are mandatory. In contrast to Cellular IP, routing changes are always initiated by the foreign domain's infrastructure.

● Transparency: HAWAII is mostly transparent to mobile nodes.

Disadvantages

● Security: There are no provisions regarding the setup of IPSec tunnels.

● Implementation: No private address support is possible because of collocated COAs.

10.Explain the advantages of Hierarchical mobile IPv6 (HMIPv6). (L-1,CO-2)

As introducing hierarchies is the natural choice for handling micro-mobility issues, several proposals for a 'hierarchical' mobile IP exist. What follows is based on Soliman, (2002). MIPv6 provides micro-mobility support by installing a **mobility anchor point (MAP)**, which is responsible for a certain domain and acts as a local HA within this domain for visiting MNs (see Figure 8.16). The MAP receives all packets on behalf of the MN, encapsulates and forwards them directly to the MN's current address (link COA, **LCOA**). As long as an MN stays within the domain of a MAP, the globally visible COA (regional COA, **RCOA**) does not change. A MAP domain's boundaries are defined by the **access routers (AR)** advertising the MAP information to the attached MNs. A MAP assists with local handovers and maps RCOA to LCOA. MNs register their RCOA with the HA using a binding update. When a MN moves locally it must only register its new LCOA with its MAP. The RCOA stays unchanged. To support smooth handovers

between MAP domains, an MN can send a binding update to its former MAP. It should be mentioned as a security benefit that mobile nodes can be provided
with some kind of limited location privacy because LCOAs on lower levels of the mobility hierarchy can be hidden from the outside world. However, this applies only to micro mobility, that is, as long as the mobile node rests in the same domain. A MN can also send a binding update to a CN who shares the same link. This reveals its location but optimizes packet flow (direct routing without going through the MAP). MNs can use their RCOA as source address. The extended mode of HMIPv6 supports both mobile nodes and mobile networks.

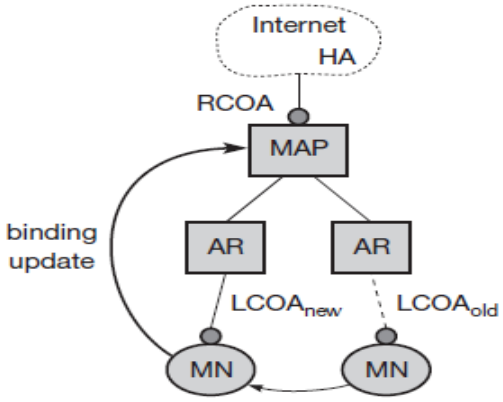


Figure 8.16
Basic architecture of hierarchical mobile IP

Advantages

● Security: MNs can have (limited) location privacy because LCOAs can be hidden.

● Efficiency: Direct routing between CNs sharing the same link is possible

Disadvantages

● Transparency: Additional infrastructure component (MAP).

● Security: Routing tables are changed based on messages sent by mobile nodes. This requires strong authentication and protection against denial of service attacks. Additional security functions might be necessary in MAPs

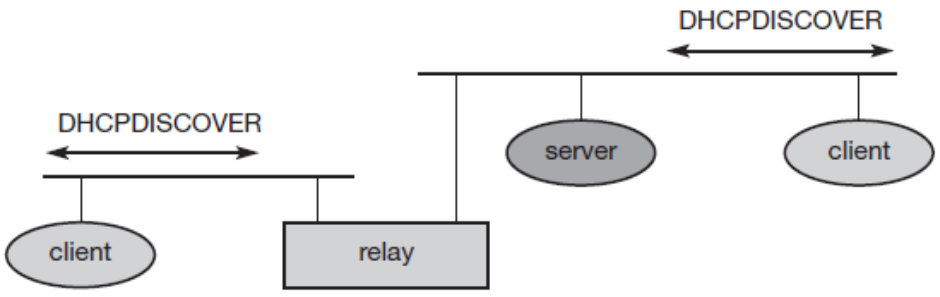
The main driving factors behind the three architectures presented here are efficiency, scalability, and seamless handover support. However, as security will be one of the key success factors of future mobile IP networks, first approaches adding this feature exist. (Mink 2000a and b.)

11.State the concepts used in Dynamic host configuration protocol techniques. (L-3,CO-2)

The dynamic host configuration protocol (DHCP, RFC 2131, Drohms, 1997) is mainly used to simplifiy the installation and maintenance of networked computers. If a new computer is connected to a network, DHCP can provide it with all the necessary information for full system integration into the network, e.g., addresses of a DNS server and the default router, the subnet mask, the domain name, and an IP address. Providing an IP address, makes DHCP very attractive for mobile IP as a source of care-of-addresses. While the basic DHCP mechanisms are quite simple, many options are available as described in RFC 2132 (Alexander, 1997).

DHCP is based on a client/server model as shown in Figure 8.17. DHCP clients send a request to a server (DHCPDISCOVER in the example) to which the server responds. A client sends requests using MAC broadcasts to reach all devices in the LAN. A DHCP relay might be needed to forward requests across inter-working units to a DHCP server.

Figure 8.17
Basic DHCP configuration



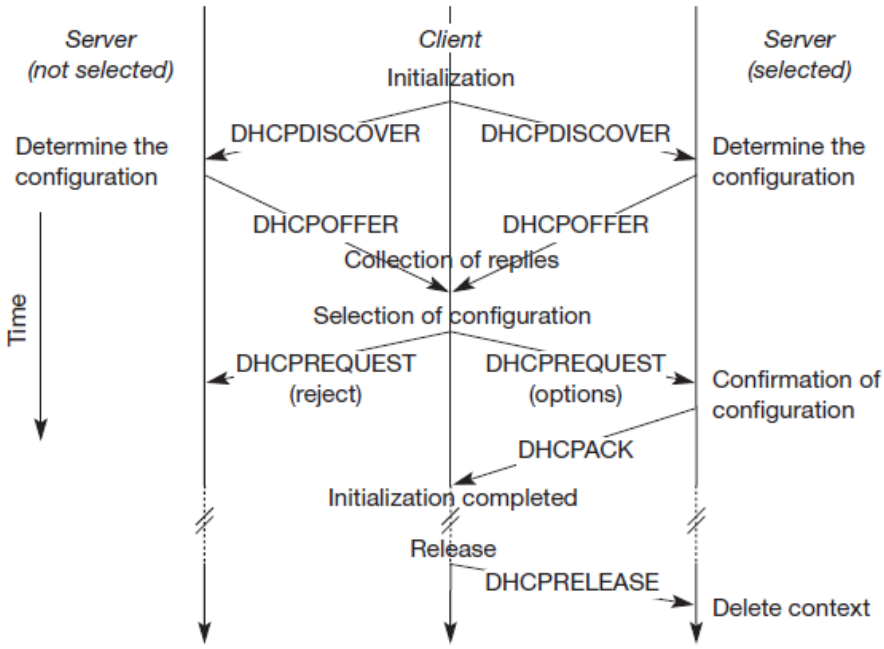


Figure 8.18
Client initialization via DHCP

A typical initialization of a DHCP client is shown in Figure 8.18. The figure shows one client and two servers. As described above, the client broadcasts a DHCPDISCOVER into the subnet. There might be a relay to forward this broadcast. In the case shown, two servers receive this broadcast and determine the configuration they can offer to the client. One example for this could be the checking of available IP addresses and choosing one for the client. Servers reply
to the client's request with DHCPOFFER and offer a list of configuration parameters. The client can now choose one of the configurations offered. The client in turn replies to the servers, accepting one of the configurations and rejecting the others using DHCPREQUEST. If a server receives a DHCPREQUEST with a rejection, it can free the reserved configuration for other possible clients. The server with the configuration accepted by the client now confirms the configuration with DHCPACK. This completes the initialization phase.

If a client leaves a subnet, it should release the configuration received by the server using DHCPRELEASE. Now the server can free the context stored for the client and

offer the configuration again. The configuration a client gets from a server is only leased for a certain amount of time, it has to be reconfirmed from time to time. Otherwise the server will free the configuration. This timeout of configuration helps in the case of crashed nodes or nodes moved away without releasing the context.

DHCP is a good candidate for supporting the acquisition of care-ofaddresses for mobile nodes. The same holds for all other parameters needed, such as addresses of the default router, DNS servers, the timeserver etc. A DHCP server should be located in the subnet of the access point of the mobile node, or at least a DHCP relay should provide forwarding of the messages. RFC 3118
(Drohms, 2001) specifies authentication for DHCP messages which is needed to protect mobile nodes from malicious DHCP servers. Without authentication, the mobile node cannot trust a DHCP server, and the DHCP server cannot trust the mobile node.

Mobile ad-hoc networks

Mobility support described in sections 8.1 and 8.2 relies on the existence of at least some infrastructure. Mobile IP requires, e.g., a home agent, tunnels, and default routers. DHCP requires servers and broadcast capabilities of the medium reaching all participants or relays to servers. Cellular phone networks require base stations, infrastructure networks etc. However, there may be several situations where users of a network cannot rely on an infrastructure, it is too expensive, or there is none at all. In these situations mobile ad-hoc networks are the only choice. It is important to note that this section focuses on so-called multi-hop ad-hoc networks when describing adhoc networking. The ad-hoc setting up of a connection with an infrastructure is not the main issue here. These networks should be mobile and use wireless communications.

Examples for the use of such mobile, wireless, multi-hop ad-hoc networks, which are only called ad-hoc networks here for simplicity, are:

- **Instant infrastructure:**

Unplanned meetings, spontaneous interpersonal communications etc. cannot rely on any infrastructure. Infrastructures need planning and administration. It would take too long to set up this kind of infrastructure; therefore, ad-hoc connectivity has to be set up.

● **Disaster relief:**

Infrastructures typically break down in disaster areas. Hurricanes cut phone and power lines, floods destroy base stations, fires burn servers. Emergency teams can only rely on an infrastructure they can set up themselves. No forward planning can be done, and the set-up must

be extremely fast and reliable. The same applies to many military activities, which is, to be honest, one of the major driving forces behind mobile ad-hoc networking research.

● **Remote areas:**

Even if infrastructures could be planned ahead, it is sometimes too expensive to set up an infrastructure in sparsely populated areas. Depending on the communication pattern, ad-hoc networks or satellite infrastructures can be a solution.

● **Effectiveness:**

Services provided by existing infrastructures might be too expensive for certain applications. If, for example, only connectionoriented cellular networks exist, but an application sends only a small status information every other minute, a cheaper ad-hoc packet-oriented network might be a better solution. Registration procedures might take too long, and communication overheads might be too high with existing networks. Application-tailored ad-hoc networks can offer a better solution.

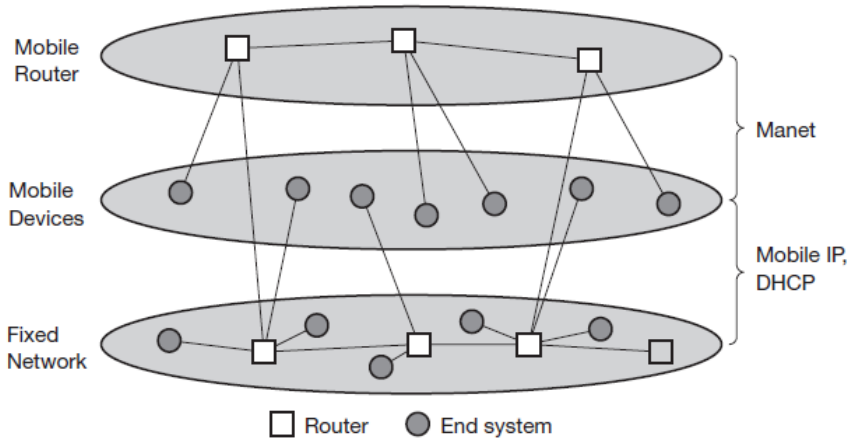


Figure 8.19
MANETs and mobile IP

Over the last few years ad-hoc networking has attracted a lot of research interest. This has led to creation of a working group at the IETF that is focusing on **mobile ad-hoc networking**, called **MANET** (MANET, 2002), (Corson, 1999). Figure 8.19 shows the relation of MANET to mobile IP and DHCP. While mobile IP and DHCP handle the connection of mobile devices to a fixed infrastructure, MANET comprises mobile routers, too. Mobile devices can be connected either
directly with an infrastructure using Mobile IP for mobility support and DHCP as a source of many parameters, such as an IP address. MANET research is responsible for developing protocols and components to enable ad-hoc networking between mobile devices. It should be noted that the separation of end system and router is only a logical separation. Typically, mobile nodes in an adhoc scenario comprise routing and end system functionality.

The reason for having a special section about ad-hoc networks within about the network layer is that routing of data is one of the most difficult issues in ad-hoc networks. General routing problems are discussed in section 8.3.1 while the following sections give some examples for routing algorithms suited to ad-hoc networks. NB: routing functions sometimes exist in layer 2, not just in the network layer (layer 3) of the reference model. Bluetooth , for example, offers forwarding/routing capabilities in layer 2 based on MAC addresses for ad-hoc networks. One of the first ad-hoc wireless networks was the packet radio network started by ARPA in 1973. It allowed up to 138

nodes in the ad-hoc network and used IP packets for data transport. This made an easy connection possible to the ARPAnet, the starting point of today's Internet. Twenty radio channels between 1718.4–1840 MHz were used offering 100 or 400 kbit/s. The system used DSSS with 128 or 32 chips/bit. A variant of distance vector routing was used in this ad-hoc network (Perlman, 1992). In this approach, each node sends a routing advertisement every 7.5 s. These advertisements contain a neighbor table with a list of link qualities to each neighbor. Each node updates the local routing table according to the distance vector algorithm based on these advertisements. Received packets also help to update the routing table. A sender now transmits a packet to its first hop neighbor using the local neighbor table. Each node forwards a packet received based on its own local neighbor table. Several enhancements to this simple scheme are needed to avoid routing loops and to reflect the possibly fast changing topology. The following sections discuss routing problems and enhanced routing mechanisms for ad-hoc networks in more detail. Perkins (2001a) comprises a collection of many routing protocols together with some initial performance considerations.

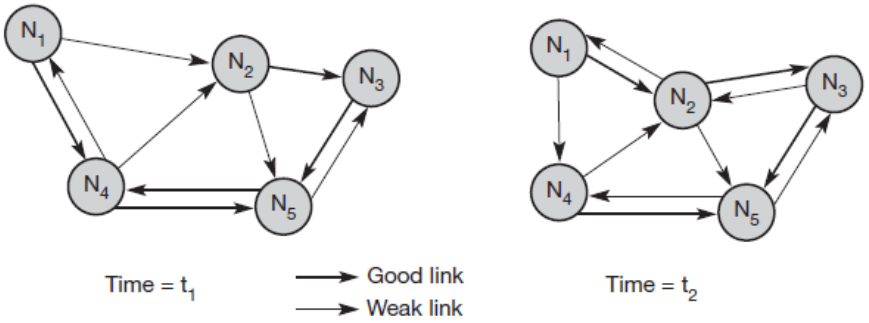
Routing

While in wireless networks with infrastructure support a base station always reaches all mobile nodes, this is not always the case in an ad-hoc network. A destination node might be out of range of a source node transmitting packets. Routing is needed to find a path between source and destination and to forward the packets appropriately. In wireless networks using an infrastructure, cells have been defined. Within a cell, the base station can reach all mobile nodes

without routing via a broadcast. In the case of ad-hoc networks, each node must be able to forward data for other nodes. This creates many additional problems that are discussed in the following paragraphs. Figure 8.20 gives a simple example of an ad-hoc network. At a certain time t1 the network topology might look as illustrated on the left side of the figure. Five nodes, N1 to N5, are connected depending on the current transmission characteristics between them. In this snapshot of the network, N4 can receive N1 over a good link, but N1 receives N4 only via a weak link. Links do not

necessarily have the same characteristics in both directions. The reasons for this are, e.g., different antenna characteristics or transmit power. N1 cannot receive N2 at all, N2 receives a signal from N1.

Figure 8.20
Example ad-hoc network



This situation can change quite fast as the snapshot at t2 shows. N1 cannot receive N4 any longer, N4 receives N1 only via a weak link. But now N1 has an asymmetric but bi-directional link to N2 that did not exist before. This very simple example already shows some fundamental differences between wired networks and ad-hoc wireless networks related to routing.

● **Asymmetric links:**

Node A receives a signal from node B. But this does not tell us anything about the quality of the connection in reverse. B might receive nothing, have a weak link, or even have a better link than the reverse direction. Routing information collected for one direction is of almost no use for the other direction. However, many routing algorithms for wired networks rely on a symmetric scenario.

● **Redundant links:**

Wired networks, too, have redundant links to survive link failures. However, there is only some redundancy in wired networks, which, additionally, are controlled by a network administrator. In ad-hoc networks nobody controls redundancy, so there might be many redundant links up to

the extreme of a completely meshed topology. Routing algorithms for wired networks can handle some redundancy, but a high redundancy can cause a large computational overhead for routing table updates.

● **Interference:**

In wired networks links exist only where a wire exists, and connections are planned by network administrators. This is not the case for wireless ad-hoc networks. Links come and go depending on transmission characteristics, one transmission might interfere with another, and nodes might overhear the transmissions of other nodes. Interference creates new problems by 'unplanned' links between nodes: if two close-by nodes forward two transmissions, they might interfere and destroy each other. On the other hand, interference might also help routing. A node can learn the topology with the help of packets it has overheard.

● **Dynamic topology:**

The greatest problem for routing arises from the highly dynamic topology. The mobile nodes might move as shown in Figure 8.20 or medium characteristics might change. This results in frequent changes in topology, so snapshots are valid only for a very short period of time. In adhoc networks, routing tables must somehow reflect these frequent changes in topology, and routing algorithms have to be adapted. Routing algorithms used in wired networks would either react much too slowly or generate too many updates to reflect all changes in topology. Routing table updates in fixed networks, for example, take place every 30 seconds. This updating frequency might be too low to be useful for ad-hoc networks. Some algorithms rely on a complete picture of the whole network. While this works in wired networks where changes are rare, it fails completely in ad-hoc networks. The topology changes during the distribution of the 'current' snapshot of the network, rendering the snapshot useless.

Let us go back to the example network in Figure 8.20 and assume that node N1 wants to send data to N3 and needs an acknowledgement. If N1 had a complete overview of the network at time t1, which is not always the case in ad-hoc networks, it would choose the path N1, N2, N3, for this requires only two hops (if we use hops as metric).

Acknowledgements cannot take the same path, N3 chooses N3, N5, N4, N1. This takes three hops and already shows that routing

also strongly influences the function of higher layers. TCP, for example, makes round trip measurements assuming the same path in both directions. This is obviously wrong in the example shown, leading to misinterpretations of measurements and inefficiencies

.

Just a moment later, at time t2, the topology has changed. Now N3 cannot take the same path to send acknowledgements back to N1, while N1 can still take the old path to N3. Although already more complicated than fixed networks, this example still assumes that nodes can have a complete insight into the current situation. The optimal knowledge for every node would be a description of the current connectivity between all nodes, the expected traffic flows, capacities of all links, delay of each link, and the computing and battery power of each node. While even in fixed networks traffic flows are not exactly predictable, for ad-hoc networks link capacities are additionally unknown. The capacity of each link can change from 0 to the maximum of the transmission technology used. In real ad-hoc networks no node knows all these factors, and establishing up-to-date snapshots of the network is almost impossible. Ad-hoc networks using mobile nodes face additional problems due to hardware limitations. Using the standard routing protocols with periodic updates wastes battery power without sending any user data and disables sleep modes. Periodic updates waste bandwidth and these resources are already scarce for wireless links.

An additional problem is interference between two or more transmissions that do not use the same nodes for forwarding. If, for example, a second transmission from node N4 to N5 (see Figure 8.20) takes place at the same time as the transmission from N1 to N3, they could interfere. Interference could take place at N2 which can receive signals from N1 and N4, or at N5 receiving N4 and N2. If shielded correctly, there is no interference between two wires.

Considering all the additional difficulties in comparison to wired networks, the following observations concerning routing can be made for ad-hoc networks with moving nodes.

● Traditional routing algorithms known from wired networks will not work efficiently (e.g., distance vector algorithms such as RIP (Hendrik, 1988), (Malkin, 1998) converge much

too slowly) or fail completely (e.g., link state algorithms such as OSPF (Moy, 1998) exchange complete pictures of the network). These algorithms have not been designed with a highly dynamic topology, asymmetric links, or interference in mind.

● Routing in wireless ad-hoc networks cannot rely on layer three knowledge alone. Information from lower layers concerning connectivity or interference can help routing algorithms to find a good path.

● Centralized approaches will not really work, because it takes too long to collect the current status and disseminate it again. Within this time the topology has already changed.

● Many nodes need routing capabilities. While there might be some without, at least one router has to be within the range of each node. Algorithms have to consider the limited battery power of these nodes.

● The notion of a connection with certain characteristics cannot work properly. Ad-hoc networks will be connectionless, because it is not possible to maintain a connection in a fast changing environment and to forward data following this connection. Nodes have to make local decisions for forwarding and send packets roughly toward the final destination.

● A last alternative to forward a packet across an unknown topology is flooding. This approach always works if the load is low, but it is very inefficient. A hop counter is needed in each packet to avoid looping, and the diameter of the ad-hoc network, i.e., the maximum number of hops, should be known. (The number of nodes can be used as an upper bound.)

Hierarchical clustering of nodes might help. If it is possible to identify certain groups of nodes belonging together, clusters can be established. While individual nodes might move faster, the whole cluster can be rather stationary. Routing between clusters might be simpler and less dynamic (see section 8.3.5.2). The following sections give two

examples for routing algorithms that were historically at the beginning of MANET research, DSDV and DSR, and useful metrics that are different from the usual hop counting. An overview of protocols follows. This is subdivided into the three categories: flat, hierarchical, and geographic-position-assisted routing based on Hong (2002).

12.How the Destination sequence distance vector used in the efficiency of the WLANs? (L-1,CO-2)

Destination sequence distance vector (DSDV) routing is an enhancement to distance vector routing for ad-hoc networks (Perkins, 1994). DSDV can be considered historically, however, an on-demand version (ad-hoc on-demand distance vector, AODV) is among the protocols currently discussed (see section 8.3.5). Distance vector routing is used as routing information protocol (RIP) in wired networks. It performs extremely poorly with certain network changes due to the count-to-infinity problem. Each node exchanges its neighbor table periodically with its neighbors. Changes at one node in the network propagate slowly through the network (step-by-step with every exchange). The strategies to avoid this problem which are used in fixed networks (poisoned-reverse/splithorizon (Perlman, 1992)) do not help in the case of wireless ad-hoc networks, due to the rapidly changing topology. This might create loops or unreachable regions within the network.

DSDV now adds two things to the distance vector algorithm:

● **Sequence numbers:**

Each routing advertisement comes with a sequence number. Within ad-hoc networks, advertisements may propagate along many paths. Sequence numbers help to apply the advertisements in correct order. This avoids the loops that are likely with the unchanged distance vector algorithm.

● **Damping:**

Transient changes in topology that are of short duration should not destabilize the routing mechanisms. Advertisements containing changes in the topology currently stored are therefore not disseminated further. A node waits with dissemination if these changes are probably unstable. Waiting time depends on the time between the first and the best announcement of a path to a certain destination.

The routing table for N1 in Figure 8.20 would be as shown in Table 8.2. For each node N1 stores the next hop toward this node, the metric (here number of hops), the sequence number of the last advertisement for this node, and the time at which the path has been installed first. The table contains flags and a settling time helping to decide when the path can be assumed stable. Router advertisements from N1 now contain data from the first, third, and fourth column: destination address, metric, and sequence number. Besides being loop-free at all times, DSDV has low memory requirements and a quick convergence via triggered updates.

Dynamic source routing

Imagine what happens in an ad-hoc network where nodes exchange packets from time to time, i.e., the network is only lightly loaded, and DSDV or one of the traditional distance vector or link state algorithms is used for updating routing tables. Although only some user data has to be transmitted, the nodes exchange routing information to keep track of the topology. These algorithms maintain routes between all nodes, although there may currently be no data exchange at all. This causes unnecessary traffic and prevents nodes from saving battery power.

Table 8.2 Part of a routing table for DSDV

Destination	Next hop	Metric	Sequence no.	Instal time
N_1	N_1	0	S_1–321	T_4–001
N_2	N_2	1	S_2–218	T_4–001
N_3	N_2	2	S_3–043	T_4–002
N_4	N_4	1	S_4–092	T_4–001
N_5	N_4	2	S_5–163	T_4–002

Dynamic source routing (DSR), therefore, divides the task of routing into two separate problems (Johnson, 1996), (Johnson, 2002a):

● **Route discovery:** A node only tries to discover a route to a destination if it has to send something to this destination and there is currently no known route.

● **Route maintenance:** If a node is continuously sending packets via a route, it has to make sure that the route is held upright. As soon as a node detects problems with the current route, it has to find an alternative.

The basic principle of source routing is also used in fixed networks, e.g. token rings. Dynamic source routing eliminates all periodic routing updates and works as follows. If a node needs to discover a route, it broadcasts a route request with a unique identifier and the destination address as parameters. Any node that receives a route request does the following.

● If the node has already received the request (which is identified using the unique identifier), it drops the request packet.

● If the node recognizes its own address as the destination, the request has reached its target.

● Otherwise, the node appends its own address to a list of traversed hops in the packet and broadcasts this updated route request. Using this approach, the route request collects a list of addresses representing a possible path on its way towards the destination. As soon as the request
reaches the destination, it can return the request packet containing the list to the receiver using this list in reverse order. One condition for this is that the links work bi-directionally. If this is not the case, and the destination node does not currently maintain a route back to the initiator of the request, it has to start a route discovery by itself. The destination may receive several lists containing different paths from the initiator. It could return the best path, the first path, or
several paths to offer the initiator a choice.

Applying route discovery to the example in Figure 8.20 for a route from N1 to N3 at time t1 results in the following.

● N1 broadcasts the request ((N1), id = 42, target = N3), N2 and N4 receive this request.

● N2 then broadcasts ((N1, N2), id = 42, target = N3), N4 broadcasts ((N1, N4), id = 42, target = N3). N3 and N5 receive N2's broadcast, N1, N2, and N5 receive N4's broadcast.

● N3 recognizes itself as target, N5 broadcasts ((N1, N2, N5), id = 42, target = N3). N3 and N4 receive N5's broadcast. N1, N2, and N5 drop N4's broadcast packet, because they all recognize an already received route request (and N2's broadcast reached N5 before N4's did).

● N4 drops N5's broadcast, N3 recognizes (N1, N2, N5) as an alternate, but longer route.

● N3 now has to return the path (N1, N2, N3) to N1. This is simple assuming symmetric links working in both directions. N3 can forward the information using the list in reverse order.

The assumption of bi-directional links holds for many ad-hoc networks. However, if links are not bi-directional, the scenario gets more complicated. The algorithm has to be applied again, in the reverse direction if the target does not maintain a current path to the source of the route request.

● N3 has to broadcast a route request ((N3), id = 17, target = N1). Only N5 receives this request.

● N5 now broadcasts ((N3, N5), id = 17, target = N1), N3 and N4 receive the broadcast.

● N3 drops the request because it recognizes an already known id. N4 broadcasts ((N3, N5, N4), id = 17, target = N1), N5, N2, and N1 receive the broadcast.

● N5 drops the request packet, N1 recognizes itself as target, and N2 broadcasts ((N3, N5, N4, N2), id = 17, target = N1). N3 and N5 receive N2's broadcast.

● N3 and N5 drop the request packet. Now N3 holds the list for a path from N1 to N3, (N1, N2, N3), and N1 knows the path from N3 to N1, (N3, N5, N4, N1). But N1 still does not know how to send data to N3! The only solution is to send the list (N1, N2, N3) with the broadcasts initiated by N3 in the reverse direction. This example shows clearly how much simpler routing can be if links are symmetrical.

The basic algorithm for route discovery can be optimized in many ways.

● To avoid too many broadcasts, each route request could contain a counter. Every node rebroadcasting the request increments the counter by one. Knowing the maximum

network diameter (take the number of nodes if nothing else is known), nodes can drop a request if the counter reaches this number.

● A node can cache path fragments from recent requests. These fragments can now be used to answer other route requests much faster (if they still reflect the topology!).

● A node can also update this cache from packet headers while forwarding other packets.

● If a node overhears transmissions from other nodes, it can also use this information for shortening routes. After a route has been discovered, it has to be maintained for as long as the
node sends packets along this route. Depending on layer two mechanisms, different approaches can be taken:

● If the link layer uses an acknowledgement (as, for example, IEEE 802.11) the node can interpret this acknowledgement as an intact route.

● If possible, the node could also listen to the next node forwarding the packet, so getting a passive acknowledgement.

● A node could request an explicit acknowledgement. Again, this situation is complicated if links are not bi-directional. If a node detects connectivity problems, it has to inform the sender of a packet, initiating a new route discovery starting from the sender. Alternatively, the node could try to discover a new route by itself. Although dynamic source routing offers benefits compared to other algorithms by being much more bandwidth efficient, problems arise if the topology
is highly dynamic and links are asymmetrical.

13.Explain the Alternative metrics used in Mobile IP layer? (L-3,CO-2)

The examples shown , typically use the number of hops as routing metric. Although very simple, especially in wireless ad-hoc networks, this is not always the best choice. Even for fixed networks, e.g., bandwidth can also be a factor for the routing metric. Due to the varying link quality and the fact that different transmissions can interfere, other metrics can be more useful. One other metric, called **least interference routing** (LIR), takes possible interference into account. Figure 8.21 shows an ad-hoc network topology. Sender S1 wants to send a packet to receiver R1, S2 to R2. Using the hop count as metric, S1 could choose three different paths with three hops, which is also the minimum. Possible paths are (S1, N3, N4, R1), (S1, N3, N2, R1), and (S1, N1, N2, R1). S2 would choose the only available path with only three hops (S2, N5, N6, R2). Taking interference into account, this picture changes. To calculate the possible

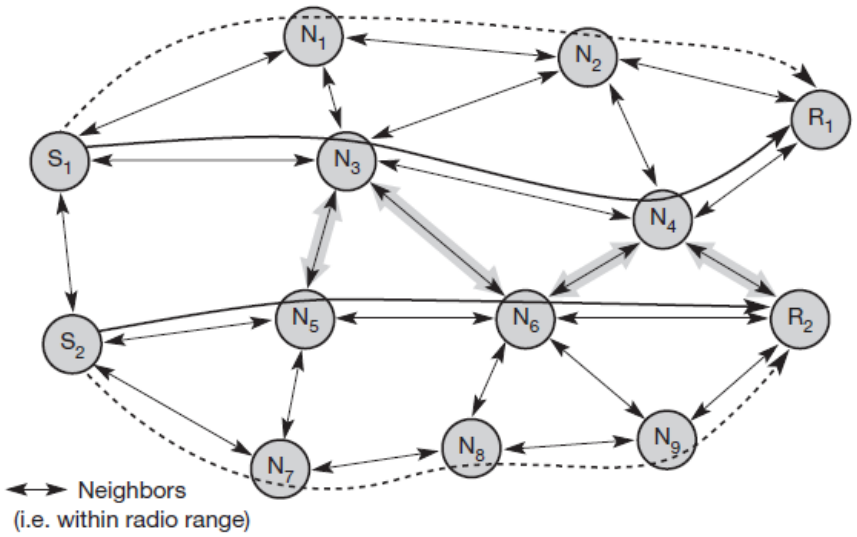


Figure 8.21
Example for least interference routing

interference of a path, each node calculates its possible interference (interference is defined here as the number of neighbors that can overhear a transmission). Every node only needs local information to compute its interference.

In this example, the interference of node N3 is 6, that of node N4 is 5 etc. Calculating the costs of possible paths between S1 and R1 results in the following:
C1 = cost(S1, N3, N4, R1) = 16,

C2 = cost(S1, N3, N2, R1) = 15,
and C3 = cost(S1, N1, N2, R1) = 12.

All three paths have the same number of hops, but the last path has the lowest cost due to interference. Thus, S1 chooses (S1, N1, N2, R1). S2 also computes the cost of different paths, examples are C4 = cost(S2, N5, N6, R2) = 16 and C5 = cost(S2, N7, N8, N9, R2) = 15. S2 would, therefore, choose the path (S2, N7, N8, N9, R2), although this path has one hop more than the first one.

With both transmissions taking place simultaneously, there would have been interference between them as shown in Figure 8.21. In this case, least interference routing helped to avoid interference. Taking only local decisions and not knowing what paths other transmissions take, this scheme can just lower the probability of interference. Interference can only be avoided if all
senders know of all other transmissions (and the whole routing topology) and base routing on this knowledge.

Routing can take several metrics into account at the same time and weigh them. Metrics could be the number of hops *h*, interference *i*, reliability *r*, error rate *e* etc. The cost of a path could then be determined as:
cost = $\alpha h + \beta i + \gamma r + \delta e + ...$

It is not at all easy (if even possible) to choose the weights α, β, γ, δ,... to achieve the desired routing behavior.

14.Brief about the Overview of ad-hoc routing protocols in detail. (L-1,CO-2)

As already mentioned, ad-hoc networking has attracted a lot of research over the last few years. This has led to the development of many new routing algorithms. They all come with special pros and cons (Royer, 1999), (Perkins, 2001a). Hong (2002)

separates them into three categories: flat routing, hierarchical routing, and geographic-position-assisted routing.

Flat ad-hoc routing

Flat ad-hoc routing protocols comprise those protocols that do not set up hierarchies with clusters of nodes, special nodes acting as the head of a cluster, or different routing algorithms inside or outside certain regions. All nodes in this approach play an equal role in routing. The addressing scheme is flat.

This category again falls into two subcategories: proactive and reactive protocols. **Proactive protocols** set up tables required for routing regardless of any traffic that would require routing functionality. DSDV, as presented in section 8.3.2 is a classic member of this group. Many protocols belonging to this group are based on a link-state algorithm as known from fixed networks. Link-state algorithms flood their information about neighbors periodically or event triggered (Kurose, 2003). In mobile ad-hoc environments this method exhibits severe drawbacks: either updating takes place often enough to reflect the actual configuration of the network or it tries to minimize network load. Both goals cannot be achieved at the same time without additional mechanisms. **Fisheye state routing** (FSR, Pei, 2000) and **fuzzy sighted link-state** (FSLS, Santivanez, 2001) attack this problem by making the update period dependent on the distance to a certain hop. Routing entries corresponding to a faraway destination are propagated with lower frequency than those corresponding to nearby destinations. The result are routing tables that reflect the proximity of a node very precisely, while imprecise entries may exist for nodes further away. Other link-state protocols that try to reduce the traffic caused by link-state information dissemination are **topology broadcast based on reverse path forwarding** (TBRPF, Ogier, 2002) and **optimized link-state routing** (OLSR, Clausen, 2002). A general **advantage** of proactive protocols is that they can give QoS guarantees related to connection set-up, latency or other realtime requirements. As long as the topology does not change too fast, the routing tables reflect the current topology with a certain precision. The propagation

characteristics (delay, bandwidth etc.) of a certain path between a sender and a receiver are already known before a data packet is sent. A big **disadvantage** of proactive schemes are their overheads in lightly loaded networks. Independent of any real communication the algorithm continuously updates the routing tables. This generates a lot of unnecessary traffic and drains the batteries of mobile devices.

Reactive protocols try to avoid this problem by setting up a path between sender and receiver only if a communication is waiting. The two most prominent members of this group are **dynamic source routing** (DSR, Johnson, 1996), as presented in section 8.3.3, and **ad-hoc on-demand distance vector** (AODV, Perkins, 2001a), an on-demand version of DSDV. AODV acquires and maintains routes only on demand like DSR does. A comparison of both protocols is given in Perkins (2001b), while Maltz (2001) gives some actual measurements done with DSR. Both protocols, DSR and AODV, are the leading candidates for standardization in the IETF. However, up to now there seems to be no clear winner. A dozen more reactive protocols already exist (Hong, 2002). A clear **advantage** of on-demand protocols is scalability as long as there is only light traffic and low mobility. Mobile devices can utilize longer low-power periods as they only have to wake up for data transmission or route discovery. However, these protocols also exhibit **disadvantages**. The initial search latency may degrade the performance of interactive applications and the quality of a path is not known *a priori*. Route caching, a mechanism typically employed by on-demand protocols, proves useless in high mobility situations as routes change too frequently

Hierarchical ad-hoc routing

Algorithms such as DSDV, AODV, and DSR only work for a smaller number of nodes and depend heavily on the mobility of nodes. For larger networks, clustering of nodes and using different routing algorithms between and within clusters can be a scalable and efficient solution. The motivation behind this approach is the locality property, meaning that if a cluster can be established, nodes typically remain within a cluster, only some change clusters. If the topology within a cluster changes, only nodes of the cluster

have to be informed. Nodes of other clusters only need to know how to reach the cluster. The approach basically hides all the small details in clusters which are further away. From time to time each node needs to get some information about the topology. Again, updates from clusters further away will be sent out less frequently compared to local updates. Clusters can be combined to form super clusters etc., building up a larger hierarchy. Using this approach, one or more nodes can act as clusterheads, representing a router for all traffic to/from the cluster. All nodes within the cluster and all other clusterheads use these as gateway for the cluster. Figure 8.22 shows an ad-hoc network with interconnection to the internet via a base station. This base station transfers data to and from the cluster heads. In this example, one cluster head also acts as head of the super cluster, routing traffic to and from the super cluster. Different routing protocols may be used inside and outside clusters. **Clusterhead-Gateway Switch Routing** (CGSR, Chiang, 1997) is a typical representative of hierarchical routing algorithms based on distance vector (DV) routing (Kurose, 2003). Compared to DV protocols, the hierarchy helps to reduce routing tables tremendously. However, it might be difficult to maintain

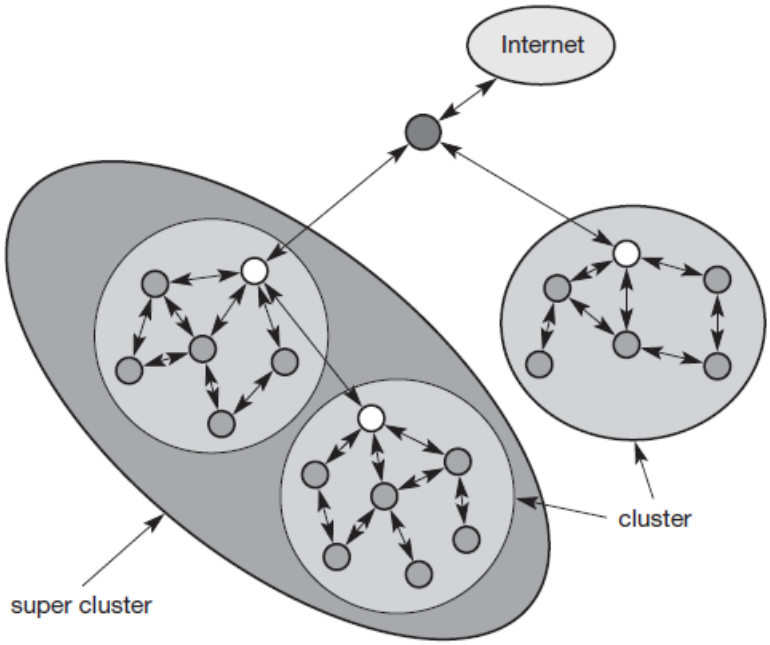


Figure 8.22
Building hierarchies in ad-hoc networks

the cluster structure in a highly mobile environment. An algorithm based on the link-state (LS) principle is **hierarchical state routing** (HSR, Pei, 1999). This applies the principle of clustering recursively, creating multiple levels of clusters and clusters of clusters etc. This recursion is also reflected in a hierarchical addressing scheme. A typical hybrid hierarchical routing protocol is the **zone routing protocol** (ZRP, Haas, 2001). Each node using ZRP has a predefined zone with the node as the center. The zone comprises all other nodes within a certain hop-limit. Proactive routing is applied within the zone, while on-demand routing is used outside the zone.

Due to the established hierarchy, HSR and CGSR force the traffic to go through certain nodes which may be a bottleneck and which may lead to suboptimal paths. Additionally, maintaining clusters or a hierarchy of clusters causes additional overheads. ZRP faces the problem of flat on-demand schemes as soon as the network size increases as many destinations are then outside the zone.

Geographic-position-assisted ad-hoc routing

If mobile nodes know their geographical position this can be used for routing purposes. This improves the overall performance of routing algorithms if geographical proximity also means radio proximity (which is typically, but not always, the case – just think of obstacles between two close-by nodes). One way to acquire position information is via the global positioning system (GPS). Mauve (2001) gives an overview of several position-based ad-hoc routing protocols. **GeoCast** (Navas, 1997) allows messages to be sent to all nodes in a specific region. This is done using addresses based on geographic information instead of logical numbers. Additionally, a hierarchy of geographical routers can be employed which are responsible for regions of different scale. The **locationaided routing** protocol (LAR, Ko, 2000) is similar to DSR, but limits route discovery to certain geographical regions. Another protocol that is based on location information is **greedy perimeter stateless routing** (GPSR, Karp, 2000). This uses only the location information of neighbors that are exchanged via periodic beacon messages or via piggybacking in data packets. The main scheme of the protocol, which is the greedy part, is quite simple. Packets are always forwarded to the neighbor that is geographically closest to the destination. Additional mechanisms are applied if a dead end is reached (no neighbor is closer to the destination than the node currently holding the data packet to be forwarded).

UNIT-III

PART A

1. **What do you meant by slow start? (L-1,CO-3)**

   TCP's reaction to a missing acknowledgement is quite drastic, but it is necessary to get rid of congestion quickly. The behavior TCP shows after the detection of congestion is called **slow start.**

2.How the fast retransmit works? (L-3,CO-3)

The gap in the packet stream is not due to severe congestion, but a simple packet loss due to a transmission error. The sender can now retransmit the missing packet(s) before the timer expires. This behaviour is called **fast retransmit**

3.**How the destination correspondent host works? (h-1,CO-3)**

Data transfer from the mobile host with **destination correspondent host** works as follows. The foreign agent snoops into the packet stream to detect gaps in the sequence numbers of TCP. As soon as the foreign agent detects a missing packet, it returns a negative acknowledgement (NACK) to the mobile host. The mobile host can now retransmit the missing packet immediately. Reordering of packets is done automatically at the correspondent host by TCP

4. **Explain about M-TCP. (L-1,CO-3)**

The **M-TCP (mobile TCP)**1 approach has the same goals as I-TCP and snooping TCP: to prevent the sender window from shrinking if bit errors or disconnection but not congestion cause current problems. M-TCP wants to improve overall throughput, to lower the delay, to maintain end-to-end semantics of TCP, and to provide a more efficient handover. Additionally, M-TCP is especially adapted to the problems arising from lengthy or frequent disconnections (Brown, 1997).

5. **Write the advantages of fast transmit/ Recovery operation? (L-3,CO-3)**

The **advantage** of this approach is its simplicity. Only minor changes in the mobile host's software already result in a performance increase. No foreign agent or correspondent host has to be changed.

6. **Write the disadvantages of fast transmit/ Recovery operation? (L-1,CO-3)**

The main **disadvantage** of this scheme is the insufficient isolation of packet losses. Forcing fast retransmission increases the efficiency, but retransmitted packets still have to cross the whole network between correspondent host and mobile host. If the handover from one foreign agent to another takes a longer time, the correspondent host will have already started retransmission.

7. **Write the advantages of Transmission/time-out freezing operation? (L-1,CO-3)**

The **advantage** of this approach is that it offers a way to resume TCP connections even after longer interruptions of the connection. It is independent of any other TCP mechanism, such as acknowledgements or sequence numbers, so it can be used together with encrypted data.

8. **Write the disadvantages of Transmission/time-out freezing operation? (L-1,CO-3)**

Not only does the software on the mobile host have to be changed, to be more effective the correspondent host cannot remain unchanged. All mechanisms rely on the capability of the MAC layer to detect future interruptions. Freezing the state of TCP does not help in case of some encryption schemes that use time-dependent random numbers. These schemes need resynchronization after interruption.

9. **Write the advantages of Selective transmission freezing operation? (L-1,CO-3)**

The **advantage** of this approach is obvious: a sender retransmits only the lost packets. This lowers bandwidth requirements and is extremely helpful in slow wireless links. The gain in efficiency is not restricted to wireless links and mobile environments. Using selective retransmission is also beneficial in all other networks.

10. **Write the advantages of Selective transmission freezing operation? (L-1,CO-3)**

However, there might be the minor **disadvantage** of more complex software on the receiver side, because now more buffer is necessary to resequence data and to wait for gaps to be filled. But while memory sizes and CPU performance permanently increase, the bandwidth of the air interface remains almost the same. Therefore, the higher complexity is no real disadvantage any longer as it was in the early days of TCP.

11.Give short notes about ECN. (L-1,CO-3)

Explicit Congestion Notification (ECN): ECN as defined in RFC 3168 allows a receiver to inform a sender of congestion in the network by setting the ECN-Echo flag on receiving an IP packet that has experienced congestion. This mechanism makes it easier to distinguish packet loss due to transmission errors from packet loss due to congestion. However, this can only be achieved when ECN capable routers are deployed in the network.

12.What happens to standard TCP in the case of disconnection? (L-2,CO-3)

A TCP sender tries to retransmit data controlled by a retransmission timer that doubles with each unsuccessful retransmission attempt, up to a maximum of one minute (the initial value depends on the round trip time). This means that the sender tries to retransmit an unacknowledged packet every minute and will give up after 12 retransmissions.

13. What happens if connectivity is back earlier than this? **(H-1,CO-3)**

No data is successfully transmitted for a period of one minute! The retransmission time-out is still valid and the sender has to wait. The sender also goes into slow-start because it assumes congestion.

14. Draw the overview of classical enhancements to TCP for mobility. (L-1,CO-3)

Approach	Mechanism	Advantages	Disadvantages
Indirect TCP	Splits TCP connection into two connections	Isolation of wireless link, simple	Loss of TCP semantics, higher latency at handover, security problems
Snooping TCP	Snoops data and acknowledgements, local retransmission	Transparent for end-to-end connection, MAC integration possible	Insufficient isolation of wireless link, security problems
M-TCP	Splits TCP connection, chokes sender via window size	Maintains end-to-end semantics, handles long term and frequent disconnections	Bad isolation of wireless link, processing overhead due to bandwidth management, security problems
Fast retransmit/ fast recovery	Avoids slow-start after roaming	Simple and efficient	Mixed layers, not transparent
Transmission/ time-out freezing	Freezes TCP state at disconnection, resumes after reconnection	Independent of content, works for longer interruptions	Changes in TCP required, MAC dependent
Selective retransmission	Retransmits only lost data	Very efficient	Slightly more complex receiver software, more buffer space needed
Transaction-oriented TCP	Combines connection setup/release and data transmission	Efficient for certain applications	Changes in TCP required, not transparent, security problems

Table 9.1 Overview of classical enhancements to TCP for mobility

PART B

1.Explain in detail about Traditional TCP and its siginificance. (L-1,CO-3)

Congestion control

A transport layer protocol such as TCP has been designed for fixed networks with fixed end-systems. Data transmission takes place using network adapters, fiber optics, copper wires, special hardware for routers etc. This hardware typically works without introducing transmission errors. If the software is mature enough, it will not drop packets or flip bits, so if a packet on its way from a sender to a receiver is lost in a fixed network, it is not because of hardware or software errors. The probable reason for a packet loss in a fixed network is a temporary overload some point in the transmission path, i.e., a state of congestion at a node. Congestion may appear from time to time even in carefully designed networks.

The packet buffers of a router are filled and the router cannot forward the packets fast enough because the sum of the input rates of packets destined for one output link is higher than the capacity of the output link. The only thing a router can do in this situation is to drop packets. A dropped packet is lost for the transmission, and the receiver notices a gap in the packet stream.Now the receiver does not directly tell the sender which packet is missing, but continues to acknowledge all in-sequence packets up to the missing one. The sender notices the missing acknowledgement for the lost packet and assumes a packet loss due to congestion. Retransmitting the missing packet and continuing at full sending rate would now be unwise, as this might only increase the congestion. Although it is not guaranteed that all packets of the
TCP connection take the same way through the network, this assumption holds for most of the packets. To mitigate congestion, TCP slows down the transmission rate dramatically. All other TCP connections experiencing the same congestion do exactly the same so the congestion is soon resolved. This cooperation of TCP connections in the internet is one of the main reasons for its survival as it is today. Using UDP is not a solution, because the throughput is higher compared to a TCP connection just at the beginning. As soon as everyone uses UDP, this advantage disappears. After that, congestion is standard and data transmission quality is unpredictable. Even under heavy load, TCP guarantees at least sharing of the bandwidth.

Slow start

TCP's reaction to a missing acknowledgement is quite drastic, but it is necessary to get rid of congestion quickly. The behavior TCP shows after the detection of congestion is called **slow start** (Kurose, 2003). The sender always calculates a **congestion window** for a receiver. The start size of the congestion window is one segment (TCP packet). The sender sends one packet and waits for acknowledgement. If this acknowledgement arrives, the sender increases the congestion window by one, now sending two packets (congestion window = 2). After arrival of the two corresponding acknowledgements, the sender again adds 2 to the congestion window, one for each of the acknowledgements. Now the congestion window equals 4. This scheme doubles the congestion window every time the acknowledgements come back, which takes one round trip time (RTT). This is called the exponential growth of the congestion window in the slow start mechanism. It is too dangerous to double the congestion window each time because the steps might become too large. The exponential growth stops at the **congestion threshold**. As soon as the congestion window reaches the congestion threshold, further increase of the transmission rate is only linear by adding 1 to the congestion window each time the acknowledgements come back. Linear increase continues until a time-out at the sender occurs due to a missing acknowledgement, or until the sender detects a gap in transmitted data because of continuous acknowledgements for the same packet. In either case the sender sets the congestion threshold to half of the current congestion window. The congestion window itself is set to one segment and the sender starts sending a single segment. The exponential growth (as described above) starts once more up to the new congestion threshold, then the window grows in linear fashion.

Fast retransmit/fast recovery

Two things lead to a reduction of the congestion threshold. One is a sender receiving continuous acknowledgements for the same packet. This informs the sender of two things. One is that the receiver got all packets up to the acknowledged packet in sequence. In TCP, a receiver sends acknowledgements only if it receives any packets from the sender. Receiving acknowledgements from a receiver also shows that the

receiver continuously receives something from the sender. The gap in the packet stream is not due to severe congestion, but a simple packet loss due to a transmission error. The sender can now retransmit the missing packet(s) before the timer expires. This behavior is called **fast retransmit** (Kurose, 2003).

The receipt of acknowledgements shows that there is no congestion to justify a slow start. The sender can continue with the current congestion window. The sender performs a **fast recovery** from the packet loss. This mechanism can improve the efficiency of TCP dramatically.

The other reason for activating slow start is a time-out due to a missing acknowledgement. TCP using fast retransmit/fast recovery interprets this congestion in the network and activates the slow start mechanism.

Implications on mobility

While slow start is one of the most useful mechanisms in fixed networks, it drastically decreases the efficiency of TCP if used together with mobile receivers or senders. The reason for this is the use of slow start under the wrong assumptions. From a missing acknowledgement, TCP concludes a congestion situation. While this may also happen in networks with mobile and wireless end-systems, it is not the main reason for packet loss.

Error rates on wireless links are orders of magnitude higher compared to fixed fiber or copper links. Packet loss is much more common and cannot always be compensated for by layer 2 retransmissions (ARQ) or error correction (FEC). Trying to retransmit on layer 2 could, for example, trigger TCP retransmission if it takes too long. Layer 2 now faces the problem of transmitting the same packet twice over a bad link. Detecting these duplicates on layer 2 is not
an option, because more and more connections use end-to-end encryption, making it impossible to look at the packet.

Mobility itself can cause packet loss. There are many situations where a soft handover from one access point to another is not possible for a mobile endsystem. For example, when using mobile IP, there could still be some packets in transit to the old foreign agent while the mobile node moves to the new foreign agent. The old foreign agent may not be able to forward those packets to the new foreign agent or even buffer the packets if disconnection of the mobile node takes too long. This packet loss has nothing to do with wireless access but is caused by the problems of rerouting traffic. The TCP mechanism detecting missing acknowledgements via time-outs
and concluding packet loss due to congestion cannot distinguish between the different causes. This is a fundamental design problem in TCP: An error control mechanism (missing acknowledgement due to a transmission error) is misused for congest on control (missing acknowledgement due to network overload). In both cases packets are lost (either due to invalid checksums or to dropping in routers). However, the reasons are completely different. TCP cannot distinguish between these two different reasons. Explicit congestion notification (ECN) mechanisms are currently discussed and some recommendations have been already given (RFC 3168, Ramakrishnan, 2001). However, RFC 3155 (Dawkins, 2001b) states that ECN cannot be used as surrogate for explicit transmission error notification. Standard TCP reacts with slow start if acknowledgements are missing, which does not help in the case of transmission errors over wireless links and which does not really help during handover. This behavior results in a severe performance degradation of an unchanged TCP if used together with wireless links or mobile nodes.

However, one cannot change TCP completely just to support mobile users or wireless links. The same arguments that were given to keep IP unchanged also apply to TCP. The installed base of computers using TCP is too large to be changed and, more important, mechanisms such as slow start keep the internet

operable. Every enhancement to TCP, therefore, has to remain compatible with the standard TCP and must not jeopardize the cautious behavior of TCP in case of

congestion. The following sections present some classical solutions before discussing current TCP tuning recommendations.

2. Brief in detail about Classical TCP improvements in the WLANs.(L-2,CO-3)

Together with the introduction of WLANs in the mid-nineties several research projects were started with the goal to increase TCP's performance in wireless and mobile environments.

Indirect TCP

Two competing insights led to the development of indirect TCP (I-TCP) (Bakre, 1995). One is that TCP performs poorly together with wireless links; the other is that TCP within the fixed network cannot be changed. I-TCP segments a TCP connection into a fixed part and a wireless part. Figure 9.1 shows an example with a mobile host connected via a wireless link and an access point to the 'wired' internet where the correspondent host resides. The correspondent node could also use wireless access. The following would then also be applied to the access link of the correspondent host.

Standard TCP is used between the fixed computer and the access point. No computer in the internet recognizes any changes to TCP. Instead of the mobile host, the access point now terminates the standard TCP connection, acting as a proxy. This means that the access point is now seen as the mobile host for the fixed host and as the fixed host for the mobile host. Between the access point and the mobile host, a special TCP, adapted to wireless links, is used. However,
changing TCP for the wireless link is not a requirement. Even an unchanged TCP can benefit from the much shorter round trip time, starting retransmission much faster. A good place for segmenting the connection between mobile host and correspondent host is at the foreign agent of mobile IP . The foreign agent controls the mobility of the mobile host anyway and can also hand over the connection to the next foreign agent when the mobile host

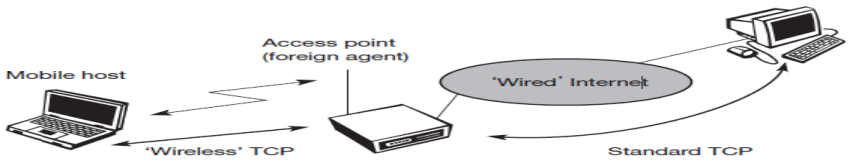


Figure 9.1
Indirect TCP segments a TCP connection into two parts

moves on. However, one can also imagine separating the TCP connections at a special server, e.g., at the entry point to a mobile phone network (e.g., IWF in GSM, GGSN in GPRS).

The correspondent host in the fixed network does not notice the wireless link or the segmentation of the connection. The foreign agent acts as a proxy and relays all data in both directions. If the correspondent host sends a packet, the foreign agent acknowledges this packet and tries to forward the packet to the mobile host. If the mobile host receives the packet, it acknowledges the
packet. However, this acknowledgement is only used by the foreign agent. If a packet is lost on the wireless link due to a transmission error, the correspondent host would not notice this. In this case, the foreign agent tries to retransmit this packet locally to maintain reliable data transport.

Similarly, if the mobile host sends a packet, the foreign agent acknowledges this packet and tries to forward it to the correspondent host. If the packet is lost on the wireless link, the mobile hosts notice this much faster due to the lower round trip time and can directly retransmit the packet. Packet loss in the wired network is now handled by the foreign agent.

I-TCP requires several actions as soon as a handover takes place. As Figure 9.2 demonstrates, not only the packets have to be redirected using, e.g., mobile IP. In the example shown, the access point acts as a proxy buffering packets for retransmission. After the handover, the old proxy must forward buffered data to the new proxy because it has already acknowledged the data. As explained , after registration with the new

foreign agent, this new foreign agent can inform the old one about its location to enable packet forwarding. Besides buffer content, the sockets of the proxy, too, must migrate to the new foreign agent located in the access point. The socket reflects the current state of the TCP connection, i.e., sequence number, addresses, ports etc. No new connection may be established for the mobile host, and the correspondent host must not see any changes in connection state.

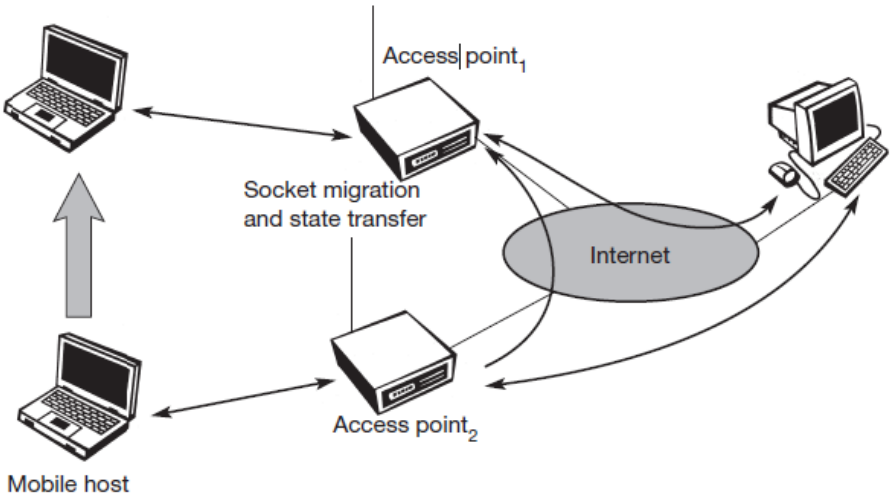


Figure 9.2 Socket and state migration after handover of a mobile host

There are several advantages with I-TCP:

● I-TCP does not require any changes in the TCP protocol as used by the hosts in the fixed network or other hosts in a wireless network that do not use this optimization. All current optimizations for TCP still work between the foreign agent and the correspondent host.

● Due to the strict partitioning into two connections, transmission errors on the wireless link, i.e., lost packets, cannot propagate into the fixed network. Without partitioning, retransmission of lost packets would take place between mobile host and correspondent host across the whole network.

Now only packets in sequence, without gaps leave the foreign agent.

● It is always dangerous to introduce new mechanisms into a huge network such as the internet without knowing exactly how they will behave. However, new mechanisms are needed to improve TCP performance (e.g., disabling slow start under certain circumstances), but with I-TCP only between the mobile host and the foreign agent. Different solutions can be tested or used at the same time without jeopardizing the stability of the internet. Furthermore, optimizing of these new mechanisms is quite simple because they only cover one single hop.

● The authors assume that the short delay between the mobile host and foreign agent could be determined and was independent of other traffic streams. An optimized TCP could use precise time-outs to guarantee retransmission as fast as possible. Even standard TCP could benefit from the short round trip time, so recovering faster from packet loss. Delay is much higher in a typical wide area wireless network than in wired networks due to FEC and MAC. GSM has a delay of up to 100 ms circuit switched, 200 ms and more packet switched (depending on packet size and current traffic). This is even higher than the delay on transatlantic links.

● Partitioning into two connections also allows the use of a different transport layer protocol between the foreign agent and the mobile host or the use of compressed headers etc. The foreign agent can now act as a gateway to translate between the different protocols. But the idea of segmentation in I-TCP also comes with some **disadvantages**:

● The loss of the end-to-end semantics of TCP might cause problems if the foreign agent partitioning the TCP connection crashes. If a sender receives an acknowledgement, it assumes that the receiver got the packet. Receiving an acknowledgement now only means (for the mobile host and a correspondent host) that the foreign agent received the packet. The correspondent node does not know anything about the partitioning, so a crashing access node may also crash applications running on the correspondent node assuming reliable end-to-end delivery.

In practical use, increased handover latency may be much more problematic. All packets sent by the correspondent host are buffered by the foreign agent besides forwarding them to the mobile host (if the TCP connection is split at the foreign agent). The foreign agent removes a packet from the buffer as soon as the appropriate acknowledgement arrives. If the mobile host now performs a handover to another foreign agent, it takes a while before the old foreign agent can forward the buffered data to the new foreign agent. During this time more packets may arrive. All these packets have to be forwarded to the new foreign agent first, before it can start forwarding the new packets redirected to it.

● The foreign agent must be a trusted entity because the TCP connections end at this point. If users apply end-to-end encryption, e.g., according to RFC 2401 (Kent, 1998a), the foreign agent has to be integrated into all security mechanisms.

Snooping TCP

One of the drawbacks of I-TCP is the segmentation of the single TCP connection into two TCP connections. This loses the original end-to-end TCP semantic. The following TCP enhancement works completely transparently and leaves the TCP end-to-end connection intact. The main function of the enhancement is to buffer data close to the mobile host to perform fast local retransmission in case of packet loss. A good place for the enhancement of TCP could be the foreign agent in the Mobile IP context (see Figure 9.3). In this approach, the foreign agent buffers all packets with **destination mobile host** and additionally ‘snoops’ the packet flow in both directions to recognize acknowledgements (Balakrishnan, 1995), (Brewer, 1998). The reason for buffering packets toward the mobile node is to enable the foreign agent to perform a local retransmission in case of packet loss on the wireless link. The foreign agent buffers every packet until it receives an acknowledgement from the mobile host. If the foreign agent does not receive an acknowledgement from the mobile host within a certain amount of time, either the packet or the acknowledgement has been lost. Alternatively,

the foreign agent could receive a duplicate ACK which also shows the loss of a packet. Now the foreign agent

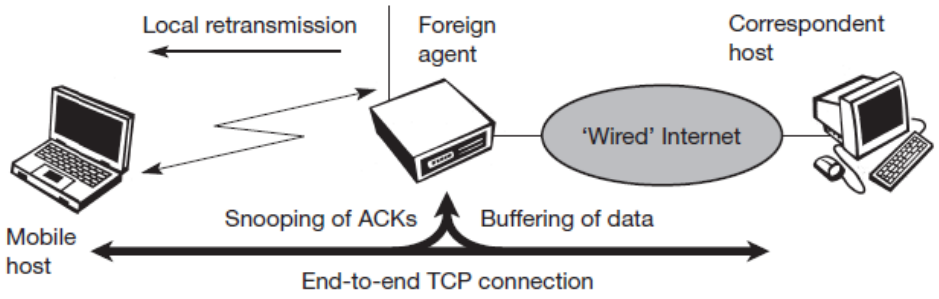


Figure 9.3 Snooping TCP as a transparent TCP extension

retransmits the packet directly from the buffer, performing a much faster retransmission compared to the correspondent host. The time out for acknowledgements can be much shorter, because it reflects only the delay of one hop plus processing time.

To remain transparent, the foreign agent must not acknowledge data to the correspondent host. This would make the correspondent host believe that the mobile host had received the data and would violate the end-to-end semantic in case of a foreign agent failure. However, the foreign agent can filter the duplicate acknowledgements to avoid unnecessary retransmissions of data from the correspondent host. If the foreign agent now crashes, the time-out of the correspondent host still works and triggers a retransmission. The foreign agent may discard duplicates of packets already retransmitted locally and acknowledged by the mobile host. This avoids unnecessary traffic on the wireless link. Data transfer from the mobile host with **destination correspondent host** works as follows. The foreign agent snoops into the packet stream to detect gaps in the sequence numbers of TCP. As soon as the foreign agent detects a missing packet, it returns a negative acknowledgement (NACK) to the mobile host. The mobile host can now retransmit the missing packet immediately. Reordering of packets is done automatically at the correspondent host by TCP.

Extending the functions of a foreign agent with a 'snooping' TCP has several **advantages**:

● The end-to-end TCP semantic is preserved. No matter at what time the foreign agent crashes (if this is the location of the buffering and snooping mechanisms), neither the correspondent host nor the mobile host have an inconsistent view of the TCP connection as is possible with I-TCP. The approach automatically falls back to standard TCP if the enhancements stop working.

● The correspondent host does not need to be changed; most of the enhancements are in the foreign agent. Supporting only the packet stream from the correspondent host to the mobile host does not even require changes in the mobile host.

● It does not need a handover of state as soon as the mobile host moves to another foreign agent. Assume there might still be data in the buffer not transferred to the next foreign agent. All that happens is a time-out at the correspondent host and retransmission of the packets, possibly already to the new care-of address.

● It does not matter if the next foreign agent uses the enhancement or not. If not, the approach automatically falls back to the standard solution. This is one of the problems of I-TCP, since the old foreign agent may have already signaled the correct receipt of data via acknowledgements to the correspondent host and now has to transfer these packets to the mobile host via the new foreign agent.

However, the simplicity of the scheme also results in some **disadvantages:**

● Snooping TCP does not isolate the behavior of the wireless link as well as ITCP. Assume, for example, that it takes some time until the foreign agent can successfully retransmit a packet from its buffer due to problems on the wireless link (congestion, interference). Although the time-out in the foreign agent may be much shorter than the one of the correspondent host, after a while the time-out in the correspondent host

triggers a retransmission. The problems on the wireless link are now also visible for the correspondent host and not fully isolated. The quality of the isolation, which snooping TCP offers, strongly depends on the quality of the wireless link, time-out values, and further traffic characteristics. It is problematic that the wireless link exhibits very high delays compared to the wired link due to error correction on layer 2 (factor 10 and more higher). This is similar to ITCP. If this is the case, the timers in the foreign agent and the correspondent host are almost equal and the approach is almost ineffective.

● Using negative acknowledgements between the foreign agent and the mobile host assumes additional mechanisms on the mobile host. This approach is no longer transparent for arbitrary mobile hosts.

● All efforts for snooping and buffering data may be useless if certain encryption schemes are applied end-to-end between the correspondent host and mobile host. Using IP encapsulation security payload (RFC 2406, (Kent, 1998b)) the TCP protocol header will be encrypted – snooping on the sequence numbers will no longer work. Retransmitting data from the foreign
agent may not work because many security schemes prevent replay attacks – retransmitting data from the foreign agent may be misinterpreted as replay. Encrypting end-to-end is the way many applications work so it is not clear how this scheme could be used in the future. If encryption is used above the transport layer (e.g., SSL/TLS) snooping TCP can be used.

3. How the Mobile TCP is playing the important role in Mobile transport layer? (L-1,CO-3)

Dropping packets due to a handover or higher bit error rates is not the only phenomenon of wireless links and mobility – the occurrence of lengthy and/or frequent disconnections is another problem. Quite often mobile users cannot connect at all. One example is islands of wireless LANs inside buildings but no coverage of the whole campus. What happens to standard TCP in the case of disconnection?A TCP sender

tries to retransmit data controlled by a retransmission timer that doubles with each unsuccessful retransmission attempt, up to a maximum of one minute (the initial value depends on the round trip time). This means that the sender tries to retransmit an unacknowledged packet every minute and will give up after 12 retransmissions. What happens if connectivity is back earlier than this? No data is successfully transmitted for a period of one minute! The retransmission time-out is still valid and the sender has to wait. The sender also goes into slow-start because it assumes congestion.

What happens in the case of I-TCP if the mobile is disconnected? The proxy has to buffer more and more data, so the longer the period of disconnection, the more buffer is needed. If a handover follows the disconnection, which is typical, even more state has to be transferred to the new proxy. The snooping approach also suffers from being disconnected. The mobile will not be able to send ACKs so, snooping cannot help in this situation.

The **M-TCP (mobile TCP)**1 approach has the same goals as I-TCP and snooping TCP: to prevent the sender window from shrinking if bit errors or disconnection but not congestion cause current problems. M-TCP wants to improve overall throughput, to lower the delay, to maintain end-to-end semantics of TCP, and to provide a more efficient handover. Additionally, M-TCP is
especially adapted to the problems arising from lengthy or frequent disconnections (Brown, 1997).

M-TCP splits the TCP connection into two parts as I-TCP does. An unmodified TCP is used on the standard host-**supervisory host (SH)** connection, while an optimized TCP is used on the SH-MH connection. The supervisory host is responsible for exchanging data between both parts similar to the proxy in ITCP (see Figure 9.1). The M-TCP approach assumes a relatively low bit error rate on the wireless link. Therefore, it does not perform caching/retransmission of data via the SH. If a packet is lost on the wireless link, it has to be retransmitted by the original sender. This maintains the TCP end-to-end semantics. The SH monitors all packets sent to the MH and ACKs returned from the

MH. If the SH does not receive an ACK for some time, it assumes that the MH is disconnected. It then chokes the sender by setting the sender's window size to 0. Setting the window size to 0 forces the sender to go into **persistent mode**, i.e., the state of the sender will not change no matter how long the receiver is disconnected. This means that the sender will not try to retransmit data. As soon as the SH (either the old SH or a new SH) detects connectivity again, it reopens the window of the sender to the old value. The sender can continue sending at full speed. This mechanism does not require changes to the sender's TCP. The wireless side uses an adapted TCP that can recover from packet loss much faster. This modified TCP does not use slow start, thus, M-TCP needs a **bandwidth manager** to implement fair sharing over the wireless link.

The **advantages** of M-TCP are the following:

● It maintains the TCP end-to-end semantics. The SH does not send any ACK itself but forwards the ACKs from the MH.

● If the MH is disconnected, it avoids useless retransmissions, slow starts or breaking connections by simply shrinking the sender's window to 0.

● Since it does not buffer data in the SH as I-TCP does, it is not necessary to forward buffers to a new SH. Lost packets will be automatically retransmitted to the new SH.

The lack of buffers and changing TCP on the wireless part also has some **disadvantages:**

● As the SH does not act as proxy as in I-TCP, packet loss on the wireless link due to bit errors is propagated to the sender. M-TCP assumes low bit error rates, which is not always a valid assumption.

● A modified TCP on the wireless link not only requires modifications to the MH protocol software but also new network elements like the bandwidth manager.

Fast retransmit/fast recovery

As described in section 9.1.4, moving to a new foreign agent can cause packet loss or time out at mobile hosts or corresponding hosts. TCP concludes congestion and goes into slow start, although there is no congestion. Section 9.1.3 showed the mechanisms of fast recovery/fast retransmit a host can use after receiving duplicate acknowledgements, thus concluding a packet loss without congestion. The idea presented by Caceres (1995) is to artificially force the fast retransmit behavior on the mobile host and correspondent host side. As soon as the mobile
host registers at a new foreign agent using mobile IP, it starts sending duplicated acknowledgements to correspondent hosts. The proposal is to send three duplicates. This forces the corresponding host to go into fast retransmit mode and not to start slow start, i.e., the correspondent host continues to send with the same rate it did before the mobile host moved to another foreign agent. As the mobile host may also go into slow start after moving to a new foreign agent, this approach additionally puts the mobile host into fast retransmit. The mobile host retransmits all unacknowledged packets using the current congestion window size without going into slow start.

The **advantage** of this approach is its simplicity. Only minor changes in the mobile host's software already result in a performance increase. No foreign agent or correspondent host has to be changed.

The main **disadvantage** of this scheme is the insufficient isolation of packet losses. Forcing fast retransmission increases the efficiency, but retransmitted packets still have to cross the whole network between correspondent host and mobile host. If the handover from one foreign agent to another takes a longer time, the correspondent host will have already started retransmission. The
approach focuses on loss due to handover: packet loss due to problems on the wireless link is not considered. This approach requires more cooperation between the mobile IP and TCP layer making it harder to change one without influencing the other.

Transmission/time-out freezing

While the approaches presented so far can handle short interruptions of the connection, either due to handover or transmission errors on the wireless link, some were designed for longer interruptions of transmission. Examples are the use of mobile hosts in a car driving into a tunnel, which loses its connection to, e.g., a satellite (however, many tunnels and subways provide connectivity via a mobile phone), or a user moving into a cell with no capacity left over. In this case, the mobile phone system will interrupt the connection. The reaction of TCP, even with the enhancements of above, would be a disconnection after a time out. Quite often, the MAC layer has already noticed connection problems, before the connection is actually interrupted from a TCP point of view. Additionally, the MAC layer knows the real reason for the interruption and does not assume congestion, as TCP would. The MAC layer can inform the TCP layer of an upcoming loss of connection or that the current interruption is not caused by congestion. TCP can now stop sending and 'freezes' the current state of its congestion window and further timers. If the MAC layer notices the upcoming interruption early enough, both the mobile and correspondent host can be informed. With a fast interruption of the wireless link, additional mechanisms in the access point are needed to inform the correspondent host of the reason for interruption. Otherwise, the correspondent host goes into slow start assuming congestion and finally breaks the connection.

As soon as the MAC layer detects connectivity again, it signals TCP that it can resume operation at exactly the same point where it had been forced to stop. For TCP time simply does not advance, so no timers expire. The **advantage** of this approach is that it offers a way to resume TCP connections even after longer interruptions of the connection. It is independent of any other TCP mechanism, such as acknowledgements or sequence numbers, so it can be used together with encrypted data. However, this scheme has some severe **disadvantages**. Not only does the software on the mobile host have to be changed, to be more effective the correspondent host cannot remain unchanged. All mechanisms rely on the capability of the MAC layer to detect

future interruptions. Freezing the state of TCP does not help in case of some encryption schemes that use time-dependent random numbers. These schemes need resynchronization after interruption.

Selective retransmission

A very useful extension of TCP is the use of selective retransmission. TCP acknowledgements are cumulative, i.e., they acknowledge in-order receipt of packets up to a certain packet. If a single packet is lost, the sender has to retransmit everything starting from the lost packet (go-back-n retransmission). This obviously wastes bandwidth, not just in the case of a mobile network, but for any network (particularly those with a high path capacity, i.e., bandwidthdelay-
product).

Using RFC 2018 (Mathis, 1996), TCP can indirectly request a selective retransmission of packets. The receiver can acknowledge single packets, not only trains of in-sequence packets. The sender can now determine precisely which packet is needed and can retransmit it.

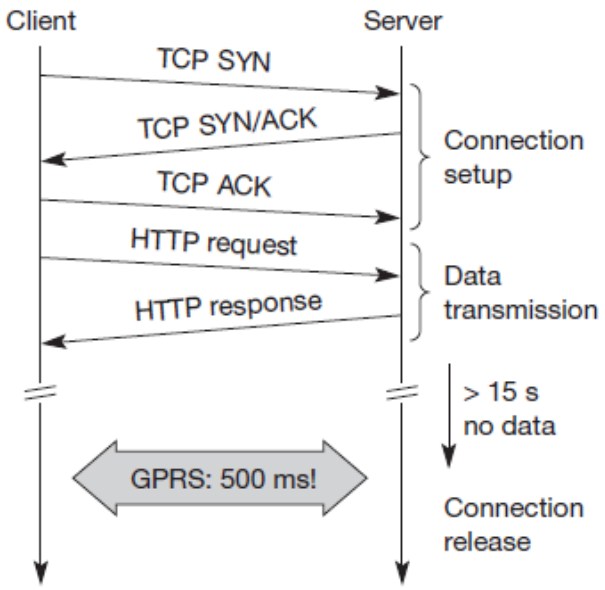
The **advantage** of this approach is obvious: a sender retransmits only the lost packets. This lowers bandwidth requirements and is extremely helpful in slow wireless links. The gain in efficiency is not restricted to wireless links and mobile environments. Using selective retransmission is also beneficial in all other networks. However, there might be the minor **disadvantage** of more complex software on the receiver side, because now more buffer is necessary to resequence data and to wait for gaps to be filled. But while memory sizes and
CPU performance permanently increase, the bandwidth of the air interface remains almost the same. Therefore, the higher complexity is no real disadvantage any longer as it was in the early days of TCP.

Transaction-oriented TCP

Assume an application running on the mobile host that sends a short request to a server from time to time, which responds with a short message. If the application requires reliable transport of the packets, it may use TCP (many applications of this kind use UDP and solve reliability on a higher, application-oriented layer). Using TCP now requires several packets over the wireless link. First, TCP uses a three-way handshake to establish the connection. At least one additional

packet is usually needed for transmission of the request, and requires three more packets to close the connection via a three-way handshake. Assuming connections with a lot of traffic or with a long duration, this overhead is minimal. But in an example of only one data packet, TCP may need seven packets altogether. Figure 9.4 shows an example for the overhead introduced by using TCP over GPRS in a web scenario. Web services are based on HTTP which requires a reliable transport system. In the internet, TCP is used for this purpose. Before a

Figure 9.4
Example TCP connection setup overhead



HTTP request can be transmitted the TCP connection has to be established. This already requires three messages. If GPRS is used as wide area transport system, one-way delays of 500 ms and more are quite common. The setup of a TCP connection already takes far more than a second.

This led to the development of a transaction-oriented TCP (T/TCP, RFC 1644 (Braden, 1994)). T/TCP can combine packets for connection establishment and connection release with user data packets. This can reduce the number of packets down to two instead of seven. Similar considerations led to the development of a transaction service in WAP.

The obvious **advantage** for certain applications is the reduction in the overhead which standard TCP has for connection setup and connection release. However, T/TCP is not the original TCP anymore, so it requires changes in the mobile host and all correspondent hosts, which is a major **disadvantage**. This solution no longer hides mobility. Furthermore, T/TCP exhibits several security problems (de Vivo, 1999).

Approach	Mechanism	Advantages	Disadvantages
Indirect TCP	Splits TCP connection into two connections	Isolation of wireless link, simple	Loss of TCP semantics, higher latency at handover, security problems
Snooping TCP	Snoops data and acknowledgements, local retransmission	Transparent for end-to-end connection, MAC integration possible	Insufficient isolation of wireless link, security problems
M-TCP	Splits TCP connection, chokes sender via window size	Maintains end-to-end semantics, handles long term and frequent disconnections	Bad isolation of wireless link, processing overhead due to bandwidth management, security problems
Fast retransmit/ fast recovery	Avoids slow-start after roaming	Simple and efficient	Mixed layers, not transparent
Transmission/ time-out freezing	Freezes TCP state at disconnection, resumes after reconnection	Independent of content, works for longer interruptions	Changes in TCP required, MAC dependent
Selective retransmission	Retransmits only lost data	Very efficient	Slightly more complex receiver software, more buffer space needed
Transaction-oriented TCP	Combines connection setup/release and data transmission	Efficient for certain applications	Changes in TCP required, not transparent, security problems

Table 9.1 Overview of classical enhancements to TCP for mobility

Table 9.1 shows an overview of the classical mechanisms presented together with some advantages and disadvantages. The approaches are not all exclusive, but can be combined. Selective retransmission, for example, can be used together with the others and can even be applied to fixed networks. An additional scheme that can be used to reduce TCP overhead is **header compression** (Degermark, 1997). Using tunneling schemes as in mobile IP (see section 8.1) together with TCP, results in protocol headers of 60 byte in case of IPv4 and 100 byte for IPv6 due to the larger addresses. Many fields in the IP and TCP header remain unchanged for every packet. Only just transmitting the differences is often sufficient. Especially delay sensitive applications like, e.g., interactive games, which have small packets benefit from small headers.

However, header compression experiences difficulties when error rates are high due to the loss of the common context between sender and receiver. With the new possibilities

of wireless wide area networks (WWAN) and their tremendous success, the focus of research has shifted more and more towards these 2.5G/3G networks. Up to now there are no final solutions to the problems arising when TCP is used in WWANs. However, some guidelines do exist.

TCP over 2.5/3G wireless networks

The current internet draft for TCP over 2.5G/3G wireless networks (Inamura, 2002) describes a profile for optimizing TCP over today's and tomorrow's wireless WANs such as GSM/GPRS, UMTS, or cdma2000. The configuration optimizations recommended in this draft can be found in most of today's TCP implementations so this draft does not require an update of millions of TCP stacks. The focus on 2.5G/3G for transport of internet data is important as already more than 1 billion people use mobile phones and it is obvious that the mobile phone systems will also be used to transport arbitrary internet data. The following characteristics have to be considered when deploying applications over 2.5G/3G wireless links:

● **Data rates:** While typical data rates of today's 2.5G systems are 10–20 kbit/s uplink and 20–50 kbit/s downlink, 3G and future 2.5G systems will initially offer data rates around 64 kbit/s uplink and 115–384 kbit/s downlink. Typically, data rates are asymmetric as it is expected that users will download more data compared to uploading. Uploading is limited by the limited
battery power. In cellular networks, asymmetry does not exceed 3–6 times, however, considering broadcast systems as additional distribution media (digital radio, satellite systems), asymmetry may reach a factor of 1,000. Serious problems that may reduce throughput dramatically are bandwidth oscillations due to dynamic resource sharing. To support multiple users within a radio cell, a scheduler may have to repeatedly allocate and deallocate resources for each user. This may lead to a periodic allocation and release of a high-speed channel.

● **Latency:** All wireless systems comprise elaborated algorithms for error correction and protection, such as forward error correction (FEC), check summing, and interleaving. FEC and interleaving let the round trip time (RTT) grow to several hundred milliseconds up to some seconds. The current GPRS standard specifies an average delay of less than two seconds for
the transport class with the highest quality .

● **Jitter:** Wireless systems suffer from large delay variations or 'delay spikes'. Reasons for sudden increase in the latency are: link outages due to temporal loss of radio coverage, blocking due to high-priority traffic, or handovers. Handovers are quite often only virtually seamless with outages reaching from some 10 ms (handover in GSM systems) to several seconds (intersystem
handover, e.g., from a WLAN to a cellular system using Mobile IP without using additional mechanisms such as multicasting data to multiple access points).

● **Packet loss:** Packets might be lost during handovers or due to corruption. Thanks to link-level retransmissions the loss rates of 2.5G/3G systems due to corruption are relatively low (but still orders of magnitude higher than, e.g., fiber connections!). However, recovery at the link layer appears as jitter to the higher layers.

Based on these characteristics, (Inamura, 2002) suggests the following configuration **parameters** to adapt TCP to wireless environments:

● **Large windows:** TCP should support large enough window sizes based on the bandwidth delay product experienced in wireless systems. With the help of the windows scale option (RFC 1323) and larger buffer sizes this can be accomplished (typical buffer size settings of 16 kbyte are not enough). A larger initial window (more than the typical one segment) of 2 to 4 segments
may increase performance particularly for short transmissions (a few segments in total).

● **Limited transmit:** This mechanism, defined in RFC 3042 (Allman, 2001) is an extension of Fast Retransmission/Fast Recovery (Caceres, 1995) and is particularly useful when small amounts of data are to be transmitted (standard for, e.g., web service requests).

● **Large MTU:** The larger the MTU (Maximum Transfer Unit) the faster TCP increases the congestion window. Link layers fragment PDUs for transmission anyway according to their needs and large MTUs may be used to increase performance. MTU path discovery according to RFC 1191 (IPv4) or RFC 1981 (IPv6) should be used to employ larger segment sizes instead of
assuming the small default MTU.

● **Selective Acknowledgement (SACK):** SACK (RFC 2018) allows the selective retransmission of packets and is almost always beneficial compared to the standard cumulative scheme.

● **Explicit Congestion Notification (ECN):** ECN as defined in RFC 3168 (Ramakrishnan, 2001) allows a receiver to inform a sender of congestion in the network by setting the ECN-Echo flag on receiving an IP packet that has experienced congestion. This mechanism makes it easier to distinguish packet loss due to transmission errors from packet loss due to congestion. However, this can only be achieved when ECN capable routers are deployed in the network.

● **Timestamp:** TCP connections with large windows may benefit from more frequent RTT samples provided with timestamps by adapting quicker to changing network conditions. With the help of timestamps higher delay spikes can be tolerated by TCP without experiencing a spurious timeout. The effect of bandwidth oscillation is also reduced.

● **No header compression:** As the TCP header compression mechanism according to RFC 1144 does not perform well in the presence of packet losses this mechanism

should not be used. Header compression according to RFC 2507 or RFC 1144 is not compatible with TCP options such as SACK or timestamps.

It is important to note that although these recommendations are still at the draft-stage, they are already used in i-mode running over FOMA as deployed in Japan and are part of the WAP 2.0 standard (aka TCP with wireless profile). 9.4 Performance enhancing proxies RFC 3135 'Performance Enhancing Proxies Intended to Mitigate Link-Related Degradations' lists many proxy architectures that can also be beneficial for wireless and mobile internet access (Border, 2001). Some initial proxy approaches, such as snooping TCP and indirect TCP have already been discussed. In principle, proxies can be placed on any layer in a communication system. However,
the approaches discussed in RFC 3135 are located in the transport and application layer. One of the key features of a proxy is its transparency with respect to the end systems, the applications and the users.

Transport layer proxies are typically used for local retransmissions, local acknowledgements, TCP acknowledgement filtering or acknowledgement handling in general. Application level proxies can be used for content filtering, content-aware compression, picture downscaling etc. Prominent examples are internet/WAP gateways making at least some of the standard web content accessible from WAP devices . Figure 9.5 shows the general architecture
of a wireless system connected via a proxy with the internet.

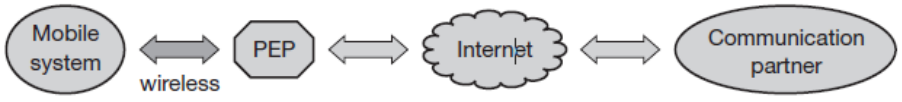


Figure 9.5 Performance enhancing proxy

However, all proxies share a common problem as they break the end-to-end semantics of a connection. According to RFC 3135, the most detrimental negative implication of breaking the end-to-end semantics is that it disables end-to-end use of IP security (RFC 2401). Using IP security with ESP (encapsulation security payload) the major part of the IP packet including the TCP header and application data is encrypted so is not

accessible for a proxy. For any application one has to choose between using a performance enhancing proxy and using IP security. This is a killer criterion in any commercial environment as the only 'solution' would mean the integration of the proxy into the security association between the end systems. Typically this is not feasible as the proxy does not belong to the same organisation as the mobile node and the corresponding node.

UNIT IV

PART A

1. What are the types of services provided by GPRS? (L-1,CO-4)

Two types of services are provided by GPRS:

Point-to-point (PTP)

Point-to-multipoint (PTM)

2. How packet data protocol helps in GPRS. (L-2,CO-4)

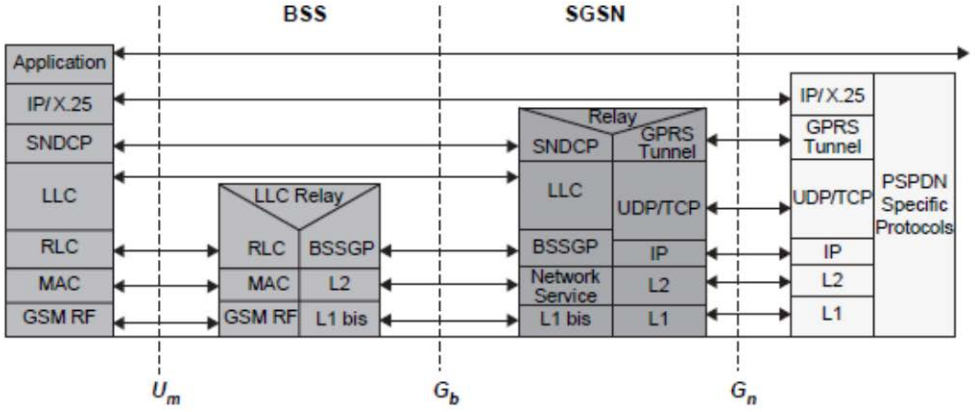
Within the GPRS networks, *protocol data units* (*PDUs*) are encapsulated at the originating GSN and decapsulated at the destination GSN. In between the GSNs, IP is used as the backbone to transfer PDUs. This whole process is referred to as tunnelling in GPRS. The GGSN also maintains routing information used to tunnel the PDUs to the SGSN that is currently serving the mobile station (MS). All GPRS user related data required by the SGSN to perform the routing and data transfer functionality is stored

within the HLR. In GPRS, a user may have multipledata sessions in operation at one time. These sessions are called *packet data protocol* (*PDP*) *contexts*.

3. Give short notes about GPRS Backbone system. (L-3,CO-4)

GPRS backbone system (*GBS*) to provide GPRS service in a similar manner to its interaction with the switching subsystem for the circuit-switched services. The GBS manages the GPRS sessions set up between the mobile terminal and the network by providing functions such as admission control, mobility management (MM), and service management (SM). Subscriber and equipment information is shared between GPRS and the switched functions of GSM by the use of a common HLR and coordination of data between the visitor location register (VLR) and the GPRS support nodes of the GBS.

4. Draw the protocol stack for GPRS. (L-1,CO-4)



SNDCP: Sub Network Dependent Convergence Protocol
BSSGP: Base Station System GPRS Protocol
LLC: Logical Link Control
RLC: Radio Link Control
MAC: Medium Access Control

Figure 15.4 Protocol stack in GPRS.

5. Give the PCCCH consists packets. (L-1,CO-4)

The PCCCH consists of:

Packet random access channel (PRACH) — uplink

Packet access grant channel (PAGCH) — downlink

Packet notifi cation channel (PNCH) — downlink

On the other hand, the PTCH can either be:

Packet data traffi c channel (PDTCH)

Packet associated control channel (PACCH)

6. What are the cases handled by mobile terminated in data ? (L-1,CO-4)

Routing to the home GPRS network, and routing to a visited GPRS network. In the fi rst case, a user sends a data packet to a mobile. The packet goes through the local area network (LAN) via a router out on the GPRS context for the mobile. If the mobile is in a GPRS idle state, the packet is rejected. If the mobile is in standby or active mode, the GGSN routes the packet in an encapsulated format to SGSN. In the second case, the home GPRS network sends the data packet over the inter-operator backbone network to the visiting GPRS network. The visiting GPRS network routes the packet to the appropriate SGSN.

7.What is meant byRadio Protocol Design? (L-1,CO-4)

The radio protocol strategy in EDGE is to reuse the protocols of GSM/GPRS whenever possible, thus minimizing the need for new protocol implementation. EDGE enhances both the GSM circuit-switched (HSCSD) and packet-switched (GPRS) mode operation. EDGE includes one packet-switched and one circuit switched mode, EGPRS and ECSD

8.Give the significance about link adaption scheme. (L-3,CO-4)

A *link adaptation* scheme regularly estimates the link quality and subsequently selects the most appropriate modulation and coding scheme for the transmission to maximize the user bit rate. The link adaptation scheme offers mechanisms for choosing the best modulation and coding scheme for the radio link.

9. Write the importance about Incremental redundancy. (H-1,CO-4)

Incremental redundancy scheme, the information is first sent with very little coding, yielding a high bit rate if decoding is immediately successful. If decoding is not

successful, additional coded bits (redundancy) are sent until decoding succeeds. The more coding that has to be sent, the lower the resulting bit rate and the higher the delay.

10. Give the short notes about ECSD. (L-1,CO-4)

Enhanced CSD (ECSD). In this case, the objective is to keep the existing GSM CS data protocols as intact as possible. In order to provide higher data rates, multislot solutions as in ECSD are provided in EDGE. This has no impact on link or system performance. A data frame is interleaved over 22 frames as in GSM, and three new 8-PSK channel coding schemes are defined along with the four already existing for GSM.

11. Draw the diagram for IMT family. (L-1,CO-4)

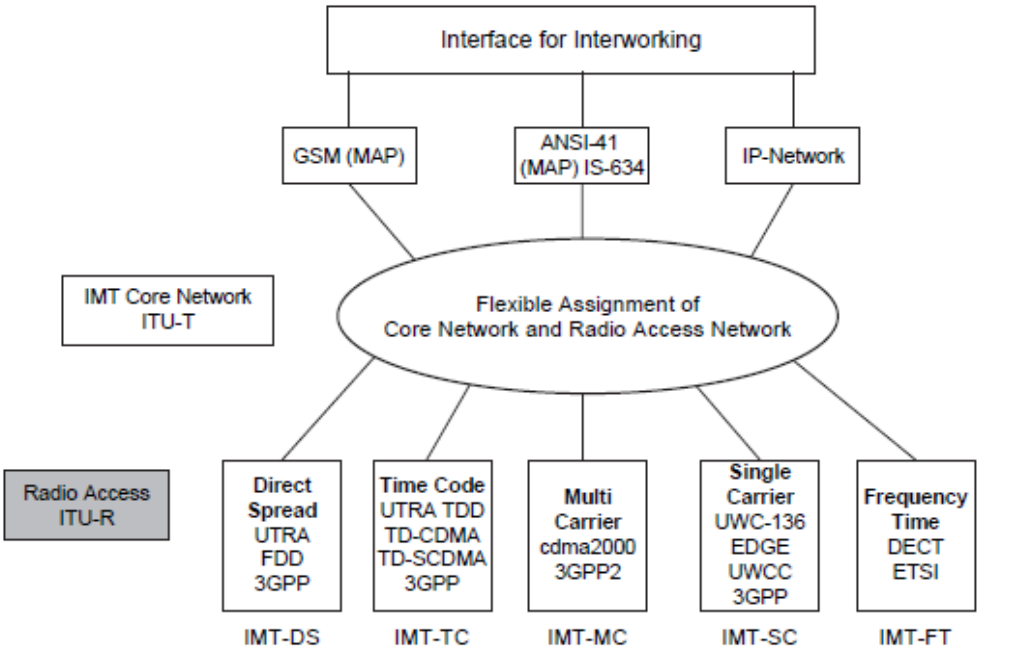


Figure 15.9 IMT family.

12. Explain *Iu, Iups, Iucs, Iur* terms in detail. (L-1,CO-4)

The *Iu* is split functionally into two logical interfaces, *Iups* connecting the packet switched domain to the access network and the *Iucs* connecting the circuit switched domain to the access network. The standards do not dictate that these are physically separate, but the user plane for each is different and the control plane may be different. The *Iur* logically connects radio network controllers (RNCs) but could be physically realized by a direct connection between RNCs or via the core network.

13. Write the services of physical layer. (L-2,CO-4)

The *physical layer* in UTRAN performs the following functions: Forward error correction, bit-interleaving, and rate matching Signal measurements, Micro-diversity distribution/combining and soft handoff execution, Multiplexing/mapping of services on dedicated physical codes

14.What are the services provide by medium access control layer? (L-3,CO-4)

The *medium access control sublayer* is responsible for efficiently transferring data for both real-time (CS) and non-real-time (PS) services to the physical layer. MAC offers services to the radio link control (RLC) sublayer and higher layers. The MAC layer provides data transfer services on logical channels

15. What does Mac responsible for? (H-1,CO-4)

MAC is responsible for

Selection of appropriate transport format (basically bit rate) within a predefined,set per information unit delivered to the physical layer, Service multiplexing on *random access channel* (*RACH*), *forward access*

channel (*FACH*), and *dedicated channel* (*DCH*)

Priority handling between data flow of a user as well as between data flows from several users - Access control on RACH and FACH, Contention resolution on RACH

16. What are the types of DPCH? (H-2,CO-4)

There are two types of DPCH: (1) *dedicated physical data channel* (*DPDCH*) to carry user data and signaling information generated at layer 2 (there may be none, one, or several DPDCHs); and (2) *dedicated physical control channel* (*DPCCH*) to carry control information generated at layer 1 (pilot bits, transmit power control (TPC) commands, feedback information (FBI) commands, and optional transport format combination indicator (TFCI))

PART B

1.Explain the GSM Evolution for Data rate for WLAN. (L-1,CO-4)

From a radio access perspective, adding 3G capabilities to 2G systems mainly means supporting higher data rates. Possible scenarios depend on spectrum availability for the network service provider. Depending on the spectrum situation, two different migration paths can be supported:

Reframing of existing spectrum bands New or modifi ed spectrum bands.

Two 3G radio access schemes are identifi ed to support the different spectrum scenarios:

1. Enhanced data rates for GSM evolution (EDGE) with high-level modulation in a 200 kHz TDMA channel is based on plug-in transceiver equipment, thereby allowing the migration of existing bands in small spectrum segments.

2. Universal mobile telecommunications services (UMTS) is a new radio access network based on 5 MHz WCDMA and optimized for efficient support of 3G services. UMTS can be used in both new and existing spectra From a network point of view, 3G capabilities implies the addition of packet switched (PS) services, Internet access, and IP connectivity. With this approach, the existing mobile networks reuse the elements of mobility support, user authentication/ service handling, and circuit switched (CS) services. With packet switched services, IP connectivity can then be added to provide a mobile multimedia core network by evolving the existing mobile network.

GSM is moving to develop enhanced cutting-edge, customer-focused solutions to meet the challenges of the new millennium and 3G mobile services [29]. When GSM was fi rst introduced, no one could have predicted the dramatic growth of the Internet and the rising demand for multimedia services. These developments have brought about new

challenges to the world of GSM. For GSM operators, the emphasis is now rapidly changing from that of instigating and driving the development of technology to fundamentally enabling mobile data transmission to that of improving speed, quality, simplicity, coverage, and reliability in terms of tools and services that will boost mass market take-up.

Users are increasingly looking to gain access to information and services wherever they are and whenever they want. GSM should provide that connectivity. Internet access, web browsing and the whole range of mobile multimedia capability are the major drivers for development of higher data speed technologies. Current data traffi c on most GSM networks is modest, less than 5% of total GSM traffi c. But with the new initiatives coming to fruition during the course of the next two to three years, exponential growth in data traffi c is forecast. The use of messaging-based applications may reach up to about 90% by the year 2008. GSM data transmission using high-speed circuit switched data (HSCSD) and GPRS may reach a penetration of about 80% by 2008 [1].

GSM operators have two nonexclusive options for evolving their networks to 3G wideband multimedia operation: (1) using GPRS and EDGE in the existing radio spectrum, and in small amounts of the new spectrum; or (2) using WCDMA in the new 2 GHz bands, or in large amounts of the existing spectrum. Both approaches offer a high degree of investment fl exibility because roll-out can proceed in line with market demand with the extensive reuse of existing network equipment and radio sites.

In the new 2 GHz bands, 3G capabilities are delivered using a new wideband radio interface that offers much higher user data rates than are available today — 384 kbps in the wide area and up to 2 Mbps in the local area. Of equal importance for such services is the high-speed packet switching provided by GPRS and its connection to public and private IP networks. GSM and digital (D)AMPS (IS-136) operators can use existing radio bands to deliver some of the 3G services, even without the new wideband spectrum by evolving current networks and deploying GPRS and EDGE technologies. In the early years of 3G service deployment, a large proportion of wireless traffi c will still be voice-only and low-rate data. So whatever the ultimate capabilities of 3G networks, effi cient and profi table ways of delivering more basic wireless

services are still needed. The signifi cance of EDGE for today's GSM operators is that it increases data rates up to 384 kbps and potentially even higher in a good quality radio environment using current GSM spectrum and carrier structures more effi ciently. EDGE is both a complement and an alternative to new WCDMA coverage. EDGE also has the effect of unifying the GSM, D-AMPS and WCDMA services through the use of dual-mode terminals.

2.How the High Speed Circuit Switched Data is performed in wireless networking techniques.. (L-2,CO-4)

High-speed circuit switched data (HSCSD) [1,4,5] is a feature that enables the co-allocation of multiple full rate traffi c channels (TCH/F) of GSM into an HSCSD confi guration. The aim of HSCSD is to provide a mixture of services with different air interface user rates by a single physical layer structure. The available capacity of an HSCSD confi guration is several times the capacity of a TCH/F, leading to a signifi cant enhancement in air interface data transfer capability. Ushering faster data rates into the mainstream is the new speed of 14.4 kbps per time slot and HSCSD protocols that approach wireline access rates of up to 57.6 kbps by using multiple 14.4 kbps time slots. The increase from the current baseline of 9.6 kbps to 14.4 kbps is due to a nominal reduction in the error-correction overhead of the GSM radio link protocol (RLP), allowing the use of a higher data rate.

For operators, migration to HSCSD brings data into the mainstream, enabled in many cases by relatively standard software upgrades to base station (BS) and mobile switching center (MSC) equipment. Flexible air interface resource allocation allows the network to dynamically assign resources related to the air interface usage according to the network operator's strategy, and the end-user's request for a change in the air interface resource allocation based on data transfer needs. The provision of the asymmetric air interface connection allows simple mobile equipment to receive data at higher rates than otherwise would be possible with a symmetric connection.

For end-users, HSCSD enables the roll-out of mainstream high-end segment services that enable faster web browsing, fi le downloads, mobile video-conference and navigation, vertical applications, telematics, and bandwidth-secure mobile local area network (LAN) access. Value-added service providers will also be able to offer

guaranteed quality of service and cost-effi cient mass-market applications, such as direct IP where users make circuit-switched data calls straight into a GSM network router connected to the Internet. To the end-user, the value-added service
provider or the operator is equivalent to an Internet service provider that offers a fast, secure dial-up Internet protocol service at cheaper mobile-to-mobile rates. HSCSD is provided within the existing mobility management. Roaming is also possible. The throughput for an HSCSD connection remains constant for the duration of the call, except for interruption of transmission during handoff. The handoff is simultaneous for all time slots making up an HSCSD connection. Endusers wanting to use HSCSD have to subscribe to general bearer services. Supplementary services applicable to general bearer services can be used simultaneously with HSCSD.Firmware on most current GSM PC cards needs to be upgraded. The reduced RLP layer also means that a stronger signal strength is necessary. Multiple time slot usage is probably only effi ciently available in off-peak times, increasing overall off-peak idle capacity usage. HSCSD is not a very feasible solution for bursty data applications.

3. Briefly explain about the techniques about General Packet Radio Service. (L-1,CO-4)

The general packet radio service (GPRS) [6,7] enhances GSM data services signifi cantly by providing end-to-end packet switched data connections. This is particularly effi cient in Internet/intranet traffi c, where short bursts of intense data communications are actively interspersed with relatively long periods of inactivity. Because there is no real end-to-end connection to be established, setting up a GPRS call is almost instantaneous and users can be continuously on-line. Users have the additional benefi ts of paying for the actual data transmitted, rather than for connection time. Because GPRS does not require any dedicated end-to-end connection, it only uses network resources and bandwidth when data is actually being transmitted. This means that a given amount of radio bandwidth can be shared efficiently among many users simultaneously.

The next phase in the high-speed road map is the evolution of current short message service (SMS), such as smart messaging and unstructured supplementary service data (USSD), toward the new GPRS, a packet data service using TCP/IP and X.25 to offer

speeds up to 115 kbps. GPRS has been standardized to optimally support a wide range of applications ranging from very frequent transmissions of medium to large data volume. Services of GPRS have been developed to reduce connection set-up time and allow an optimum usage of radio resources. GPRS provides a packet data service for GSM where time slots on the air interface can be assigned to GPRS over which packet data from several mobile stations is multiplexed.

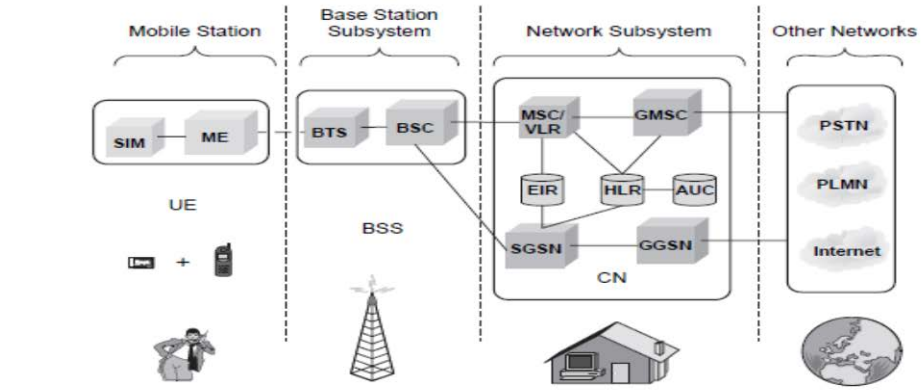
A similar evolution strategy, also adopting GPRS, has been developed for DAMPS (IS-136). For operators planning to offer wideband multimedia services, the move to GPRS packet-based data bearer service is significant; it is a relatively small step compared to building a totally new 3G IMT-2000 network. Use of the GPRS network architecture for IS-136_ packet data service enables data subscription roaming with GSM networks around the globe that support GPRS and

its evolution. The IS-136_ packet data service standard is known as GPRS-136. GPRS-136 provides the same capabilities as GSM GPRS. The user can access either X.25 or an IP-based data network. only to expand the wireless data market in preparation for the introduction of 3G services, but also a platform on which to build IMT-2000 frequencies should they acquire them.

The implementation of GPRS has a limited impact on the GSM core network .

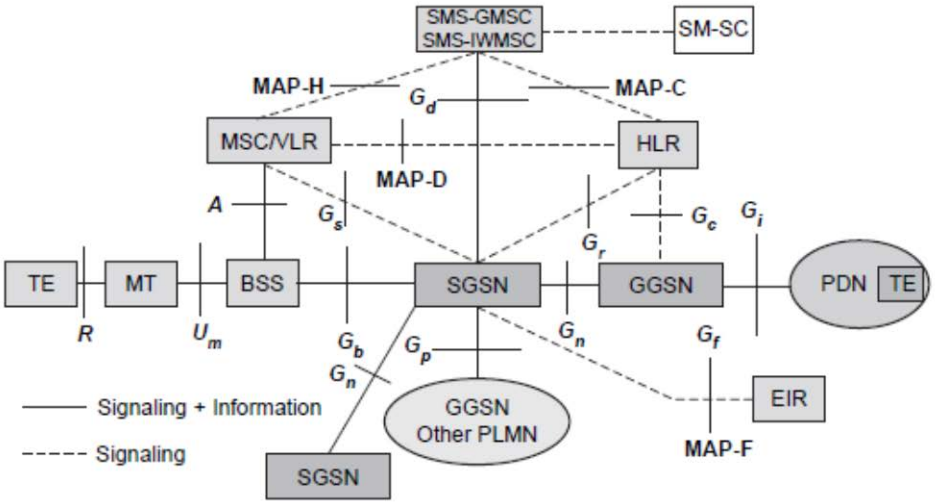
It simply requires the addition of new packet data switching and gateway nodes, and an upgrade to existing nodes to provide a routing path for packet data between the wireless terminal and a gateway node. The gateway node provides interworking with external packet data networks for access to the Internet, intranet, and databases.

A GPRS architecture for GSM is shown in Figure 15.2 and network element interfaces in Figure 15.3. GPRS supports all widely used data communications



SGSN: Serving GPRS Support Node
GGSN: Gateway GPRS Support Node
HLR: Home Location Register
VLR: Visitor Location Register
MSC: Mobile Switching Center
BSS: Base Station System
GMSC: Gateway MSC
EIR: Equipment Identity Register
ME: Mobile Equipment
SIM: Subscriber Identity Card
PLMN: Public Land Mobile Network

Figure 15.2 A GPRS architecture in GSM.



SGSN: Serving GPRS Support Node
GGSN: Gateway GPRS Support Node
MAP: Mobile Application Part
HLR: Home Location Register
VLR: Visitor Location Register
MSC: Mobile Switching Center
BSS: Base Station System
GMSC: Gateway MSC
IWMSC: Interworking MSC
TE: Terminal Equipment
MT: Mobile Terminal
EIR: Equipment Identity Register
SMS: Short Message Service

Figure 15.3 GPRS interfaces for different network elements.

protocols, including IP, so it is possible to connect with any data source from anywhere in the world using a GPRS mobile terminal. GPRS supports applications ranging from

low-speed short messages to high-speed corporate LAN communications. However, one of the key benefi ts of GPRS — that it is connected through the existing GSM air interface modulation scheme — is also a limitation, restricting its potential for delivering higher data rates than 115 kbps. To build even higher rate data capabilities into GSM, a new modulation scheme is needed.

GPRS can be implemented in the existing GSM systems. Changes are required in an existing GSM network to introduce GPRS. The base station subsystem (BSS) consists of a base station controller (BSC) and packet control unit (PCU). The PCU supports all GPRS protocols for communication over the air interface. Its function is to set up, supervise, and disconnect packet switched calls. The packet control unit supports cell change, radio resource confi guration, and channel assignment. The base station transceiver (BTS) is a relay station without protocol functions. It performs modulation and demodulation. The GPRS standard introduces two new nodes, the *serving GPRS support node* (*SGSN*) and the *gateway GPRS support node* (*GGSN*). The home location register (HLR) is enhanced with GPRS subscriber data and routing information. Two types of services are provided by GPRS:

Point-to-point (PTP)

Point-to-multipoint (PTM)

Independent packet routing and transfer within the public land mobile network (PLMN) is supported by a new logical network node called the *GPRS support node* (*GSN*). The GGSN acts as a logical interface to external packet data networks. Within the GPRS networks, *protocol data units* (*PDUs*) are encapsulated at the originating GSN and decapsulated at the destination GSN. In between the GSNs, IP is used as the backbone to transfer PDUs. This whole process is referred to as tunnelling in GPRS. The GGSN also maintains routing information used to tunnel the PDUs to the SGSN that is currently serving the mobile station (MS). All GPRS user related data required by the SGSN to perform the routing and data transfer functionality is stored within the HLR. In GPRS, a user may have multipledata sessions in operation at one time. These sessions are called *packet data protocol* (*PDP*) *contexts*. The number of PDP contexts that are open for a user is only limited by the user's subscription and any operational constraints of the network. The main goal of the GPRS-136 architecture is to integrate IS-136 and

GSM GPRS as much as possible with minimum changes to both technologies. In order to provide subscription roaming between GPRS-136 and GSM GPRS networks, a separate functional GSM GPRS HLR is incorporated into the architecture in addition to the IS-41 HLR. The European Telecommunication Standards Institute (ETSI) has specifi ed GPRS as an overlay to the existing GSM network to provide packet data services.In order to operate a GPRS over a GSM network, new functionality has been introduced into existing GSM network elements (NEs) and new NEs are integrated into the existing service provider's GSM network.

The BSS of GSM is upgraded to support GPRS over the air interface. The BSS works with the *GPRS backbone system* (*GBS*) to provide GPRS service in a similar manner to its interaction with the switching subsystem for the circuit-switched services. The GBS manages the GPRS sessions set up between the mobile terminal and the network by providing functions such as admission control, mobility management (MM), and service management (SM). Subscriber and equipment information is shared between GPRS and the switched functions of GSM by the use

of a common HLR and coordination of data between the visitor location register (VLR) and the GPRS support nodes of the GBS. The GBS is composed of two new NEs — the SGSN and the GGSN. The SGSN serves the mobile and performs security and access control functions. The SGSN is connected to the BSS via frame-relay. The SGSN prov packet routing, mobility management, authentication, and ciphering to and from all GPRS subscribers located in the SGSN service area. A GPRS subscriber may be served by any SGSN in the network, depending on location. The traffi c is routed from the SGSN to the BSC and to the mobile terminal via a BTS. At GPRS *attach*, the SGSN establishes a mobility management context containing information about mobility and security for the mobile. At PDP context activation, the SGSN establishes a PDP context which is used for routing purposes with the GGSN that the GPRS subscriber uses. The SGSN may send in some cases location information to the MSC/VLR and receive paging requests. The GGSN provides the gateway to the external IP network, handling security and accounting functions as well as the dynamic allocation of IP addresses.The GGSN contains routing information for the attached GPRS users. The routing information is used to tunnel PDUs to the mobile's current point of attachment,

SGSN. The GGSN may be connected with the HLR via optional interface Gc. The GGSN is the fi rst point of public data network (PDN) interconnection with a GSM PLMN supporting GPRS. From the external IP network's point of view, the GGSN is a host that owns all IP addresses of all subscribers served by the GPRS network.

The PTM-SC handles PTM traffic between the GPRS backbone and the HLR. The nodes are connected by an IP backbone network. The SGSN and GGSN functions may be combined in the same physical node or separated — even residing in different mobile networks.

A special interface (Gs) is provided between MSC/VLR and SGSN to coordinate signaling for mobile terminals that can handle both circuit-switched and packet-switched data.

The HLR contains GPRS subscription data and routing information, and can be accessible from the SGSN. For the roaming mobiles, the HLR may reside in a different PLMN than the current SGSN. The HLR also maps each subscriber to one or more GGSNs.

The objective of the GPRS design is to maximize the use of existing GSM infrastructure while minimizing the changes required within GSM. The GSN contains most of the necessary capabilities to support packet transmission over GSM. The critical part in the GPRS network is the mobile to GSN (MS-SGSN) link which includes the MS-BTS, BTS-BSC, BSC-SGSN, and the SGSN-GGSN link. In particular, the Um interface including the radio channel is the bottleneck of the GPRS network due to the spectrum and channel speed/quality limitations. Since multiple traffi c types of varying priorities are supported by the GPRS network, the quality of service criteria as well as resource management is required for performance evaluation.

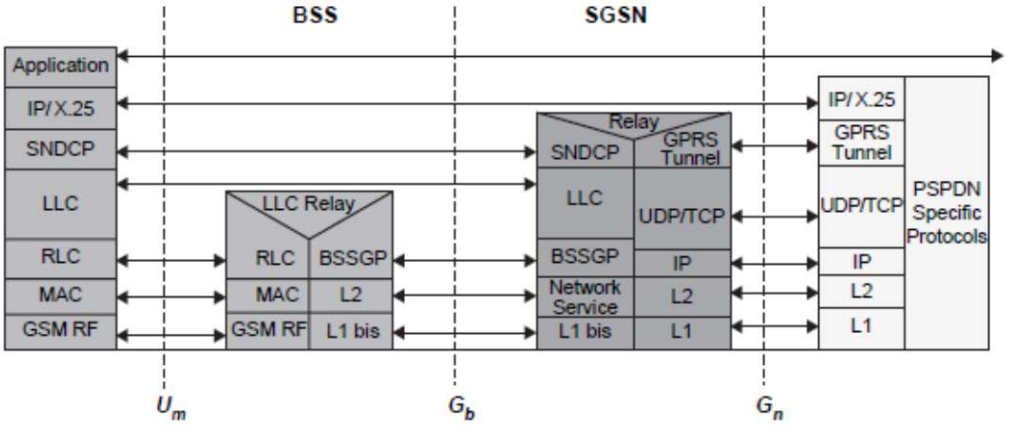
The BSC will require new capabilities for controlling the packet channels, new hardware in the form of a packet control unit, and new software for GPRS mobility management and paging. The BSC also has a new traffi c and signaling interface from the SGSN. The BTS has new protocols supporting packet data for the air interface, together with new slot and channel resource allocation functions. The utilization of resources is optimized through dynamic sharing between the two traffi c types handled by the BSC.

MS-SGSN Link

The logical link control (LLC) layer (see Figure 15.4) is responsible for providing a link between the MS and the SGSN. It governs the transport of GPRS signaling and traffi c information from the MS to the SGSN. GPRS supports three service access points (SAPs) entities: the layer 3 management, subnet dependent convergence, and short message service (SMS). On the MS-BSS link, the radio link control (RLC), the medium access control (MAC), and GSM RF protocols

are supported. The main drawback in implementing GPRS on an existing GSM infrastructure

is that the GSM network is optimized for voice transmission (i.e., the GSM channel quality is designed for voice which can tolerate errors at a predefi ned level). It is therefore expected that GPRS could have varied transmission performance in a different network or coverage area. To overcome this problem, GPRS supports multiple coding rates at the physical layer. A GPRS could share radio resources with GSM circuit switched (CS) service. This is governed by a dynamic resource sharing based on the capacity of demand criteria. A GPRS channel is allocated only if an active GPRS terminal exists in the



SNDCP: Sub Network Dependent Convergence Protocol
BSSGP: Base Station System GPRS Protocol
LLC: Logical Link Control
RLC: Radio Link Control
MAC: Medium Access Control

Figure 15.4 Protocol stack in GPRS.

network. Once resources are allocated to GPRS, at least one channel will serve as the *master* channel to carry all necessary signaling and control information for the operation of the GPRS. All other channels will serve as *slave* and are only used to carry user and signaling information. If no master channel exists, all the GPRS users will use the GSM *common control channel* (*CCCH*) and inform the network to allocate GPRS resources.

A physical channel dedicated to GPRS is called a *packet data channel* (*PDCH*). It is mapped into one of the physical channels allocated to GPRS (see Figure 15.5). A PDCH can either be used as a *packet common control channel* (*PCCCH*), a *packet broadcast control channel* (*PBCCH*), or a *packet traffic channel* (*PTCH*).

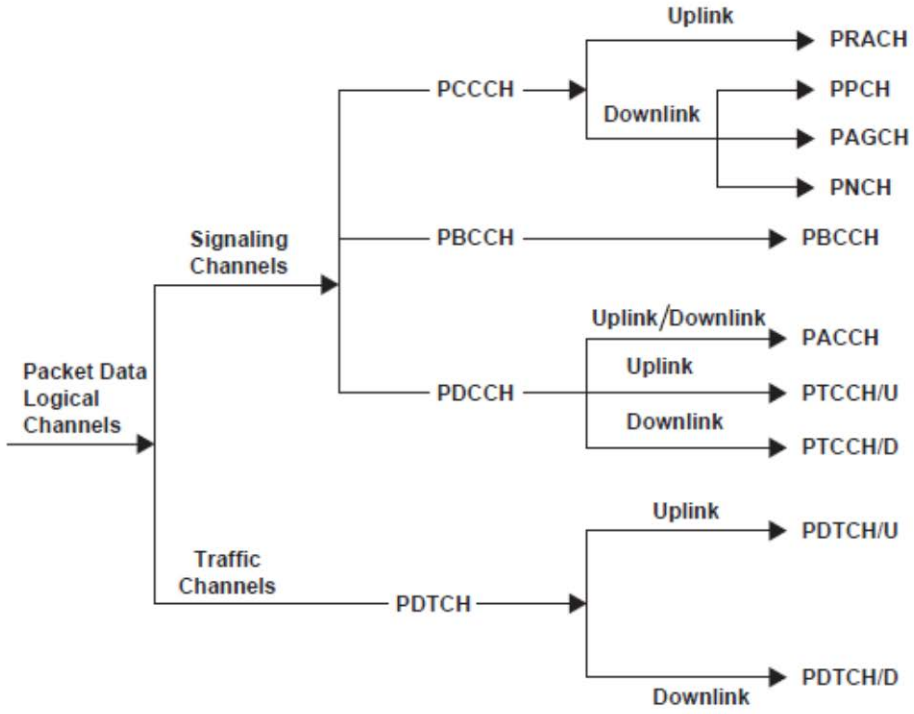


Figure 15.5 GPRS logical channels.

The PCCCH consists of:

Packet random access channel (PRACH) — uplink
Packet access grant channel (PAGCH) — downlink
Packet notifi cation channel (PNCH) — downlink
On the other hand, the PTCH can either be:
Packet data traffi c channel (PDTCH)
Packet associated control channel (PACCH)
The arrangement of GPRS logical channels for given traffi c characteristics also requires the combination of PCCCHs and PTCHs. Fundamental questions such as how many PDTCHs can be supported by a single PCCCH is needed in dimensioning GPRS.

RLC/MAC Layer

The multiframe structure of the PDCH in which GPRS RLC messages are transmitted is composed of 52 TDMA frames organized into RLC blocks of four bursts resulting in 12 blocks per multiframe plus four idle frames located in the 13th, 26th, 39th, and 52nd positions (see Figure 15.6). B0 consists of frames 1, 2, 3 and 4, B1 consists of frames 5, 6, 7, and 8 and so on. It is important that the mapping of logical channels onto the radio blocks is done by means of an ordered set of blocks (B0, B6, B9, B1, B7, B4, B10, B2, B8, B5, B11). The advantage of ordering the blocks is mainly to spread the locations of the control channels in each time slot reducing the average waiting time for the users to transmit signaling packets. It also provides an interleaving of the GPRS multiframe.

GPRS uses a reservation protocol at the MAC layer. Users that have packets ready to send request a channel via the PRACHs. The random access burst consists of only one TDMA frame with duration enough to transmit an 11-bit signaling message. Only the PDCHs carrying PCCCHs contain PRACHs. The blocks used as PRACHs are indicated by an uplink state fl ag (USF _ free) by the downlink pair channel.

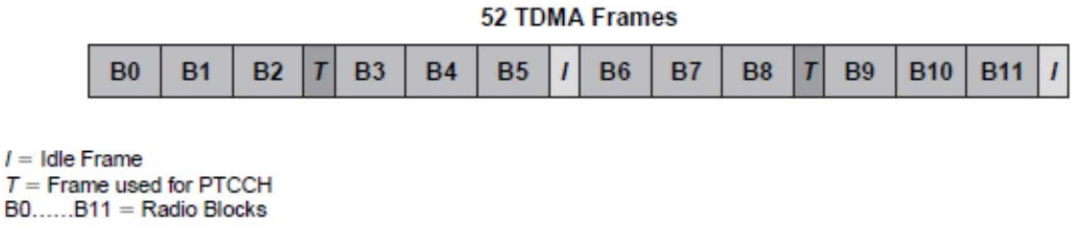


Figure 15.6 Idle frame location in GPRS multiframe.

Alternatively, the fi rst *K* blocks following the ordered set of blocks can be assigned to PRACH permanently. The access burst is transmitted in one of the four bursts assigned as PRACH. Any packet channel request is returned by a packet immediate assignment on the PRACHs whose locations are broadcast by PBCCH. Optionally, a packet resource request for additional channels is initiated and returned by a packet resource assignment. The persistence of random access is maintained by the traffic load and user class with a back-off algorithm for unsuccessful attempts. In the channel assignment, one or more PTCHs (time slot) will be allocated to a particular user. A user reserves a specifi c number of blocks on the assigned PTCH as indicated by the USF. It is possible to accommodate more than one user per PTCH. User signaling is also transmitted on the same PTCH using the PAGCH whose usage depends on the necessity of the user.

The performance of the MAC layer depends on the logical arrangement of the GPRS channels (i.e., allocation of random access channels, access grant channels, broadcast channels, etc.) for given traffi c statistics. This is determined by the amount of resources allocated for control and signaling compared to data traffi c. A degree of fl exibility of logical channels is also achieved as the traffi c varies. The arrangement of logical channels is determined through the PBCCH.

LLC Layer

The LLC layer is responsible for providing a reliable link between the mobile and the SGSN. It is based on the LAPD (link access protocol D) protocol. It is designed to support variable length transmission in a PTP or PTM topology. It includes the layer function such as sequence control, fl ow control, error detection, ciphering, and recovery as well as the provision of one or more logical link connections between two layer 3 entities. A logical link is identifi ed by a DLCI (data link control identity) which consists of a service access point identity (SAPI) and terminal equipment identity (TEI) mapped on the LLC frame format. Depending on the status of the logical link, it supports an unacknowledged or an acknowledged information transfer. The former does not support error recovery mechanisms. The acknowledged information transfer supports error and flow control. This operation only applies to point-to-point operations. The LLC frame consists of an address field (1 or 5 octets), control fi eld (2 or 6 octets), a length indicator fi eld (2 octets maximum), information fi elds (1500 octets maximum), and a frame check sequence of 3 octets. Four types of control fi eld formats are allowed including the supervisory functions (S format), the control functions (U), and acknowledged and unacknowledged information transfer (I and UI). In the performance evaluation, the objective is to determine delay during the exchange of commands and responses involved in various operations supported by the LLC in relation to the transfer of an LLC PDU. The LLC commands and responses are exchanged between two layer 3 entities in conjunction with a service primitive invoked by the mobile or the SGSN.

Data Packet Routing in the GPRS Network

The following discusses data packet routing for the mobile originated and mobile terminated data call scenarios. In mobile originated data routing, the mobile gets an IP packet from an application and requests a channel reservation. The mobile transmits data in the reserved time slots. The packet switched public data network (PSPDN) PDU is encapsulated into a sub-network dependent convergence protocol (SNDCP) unit that is sent via LLC protocol over the air interface to the SGSN currently serving the mobile (see Figure 15.7).

For mobile terminated data routing (see Figure 15.7), we have two cases:

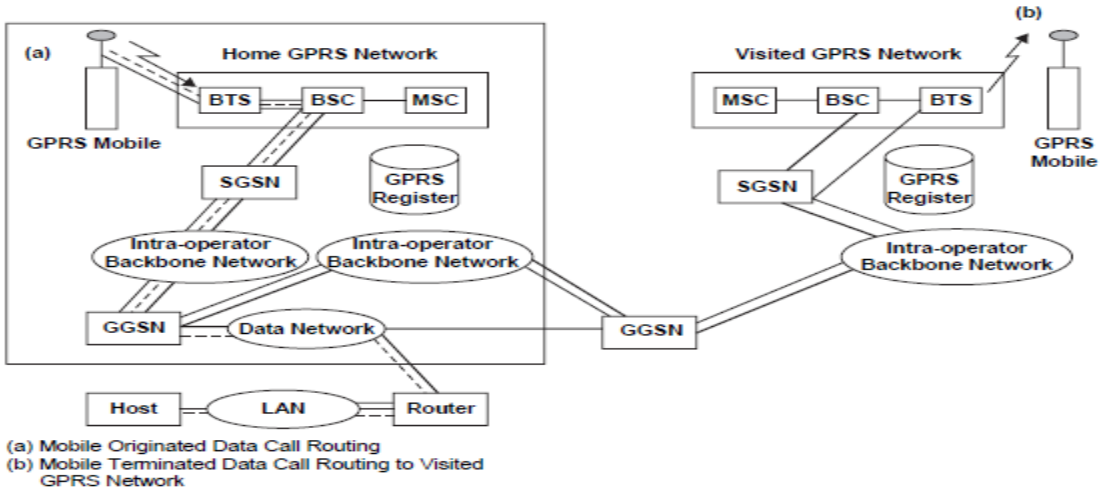
routing to the home GPRS network, and routing to a visited GPRS network. In the fi rst case, a user sends a data packet to a mobile. The packet goes through the local area network (LAN) via a router out on the GPRS context for the mobile. If the mobile is in a GPRS idle state, the packet is rejected. If the mobile is in standby or active mode, the GGSN routes the packet in an encapsulated format to SGSN. In the second case, the home GPRS network sends the data packet over the inter-operator backbone network to the visiting GPRS network. The visiting GPRS network routes the packet to the appropriate SGSN.

The PTP and PTM applications of GPRS are listed below:

Point-to-point

Messaging (e.g., e-mail)

Remote access to corporate networks



(a) Mobile Originated Data Call Routing
(b) Mobile Terminated Data Call Routing to Visited GPRS Network

Access to the Internet

Credit card validation (point-of-sales)

Utility meter readings

Road toll applications

Automatic train control

Point-to-multipoint

PTM-multicast (send to all)

News

Traffi c information

Weather forecasts

Financial updates

PTM-group call (send to some)

Taxi fl eet management

Conferencing

GPRS provides a service for bursty and bulky data transfer, radio resources on demand, shared use of physical radio resources, existing GSM functionality, mobile applications for a mass application market, volume dependent charging, and integrated services, operation and management.

4.Explain the Enhanced Data Rates for GSM Enhancement in networking. (L-1,CO-4)

The enhanced data rates for GSM enhancements (EDGE) [8] provides an evolutionary path from existing 2G systems (GSM, IS-136, PDC) to deliver some 3G services in existing spectrum bands. The advantages of EDGE include fast availability, reuse of existing GSM, IS-136 and PDC infrastructure, as well as support for the gradual introduction of 3G capabilities.

EDGE is primarily a radio interface improvement, but it can also be viewed as a system concept that allows GSM, IS-136, and PDC networks to offer a set of new services. EDGE has been designed to improve S/I by using link quality control. Link quality control adapts the protection of the data to the channel quality so that for all channel qualities an optimal bit rate is achieved.

EDGE can be seen as a generic air interface for effi ciently providing high bit rates, facilitating an evolution of existing 2G systems toward 3G systems. The EDGE air interface is designed to facilitate higher bit rates than those currently achievable in existing 2G systems. The modulation scheme based on 8-PSK is used to increase the gross bit rate. GMSK modulation as defi ned in GSM is also part of the EDGE system. The symbol rate is 271 ksps for both GMSK and 8-PSK,

leading to gross bit rates per time slot of 22.8 kbps and 69.2 kbps, respectively.

The 8-PSK pulse shape is linearized by GMSK to allow 8-PSK to fi t into the GSM

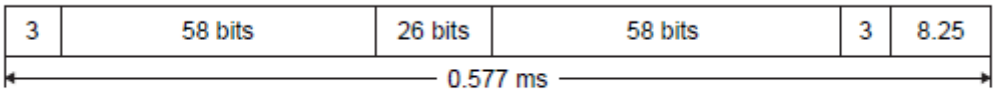


Figure 15.8 Burst format for EDGE with 8-PSK.

spectrum mask. The 8-PSK burst format is similar to GSM (see Figure 15.8). EDGE reuses the GSM carrier bandwidth and time slot structure. EDGE (also known as the 2.5G system) has been designed to enhance user bandwidth through GPRS. This is achieved through the use of higher-level modulation schemes. Although EDGE reuses the GSM carrier bandwidth and time slot structure, the technique is by no means restricted to GSM systems; it can be used as a generic air interface for effi cient provision of higher bit rates in other TDMA systems. In the Universal Wireless Communications Consortium (UWCC) the 136 high speed (136 HS) radio transmission technology (RTT) radio interface was proposed as a means to satisfy the requirements for an IMT-2000 RTT. EDGE was adopted by UWCC in 1998 as the outdoor component of 136 HS to

provide 384 kbps data service.

The standardization effort for EDGE has two phases. In the fi rst phase of EDGE the emphasis has been placed on enhanced GPRS (EGPRS) and enhanced CSD (ECSD). The second phase is defi ned with improvements for multimedia and real-time services.

In order to achieve a higher gross rate, new modulation scheme, quaternary offset quadrature amplitude modulation (QOQAM) has been proposed for EDGE, since it can provide higher data rates and good spectral effi ciency. An offset modulation scheme is proposed because it gives smaller amplitude variation than 16-QAM, which can be benefi cial when using nonlinear amplifi ers. EDGE co-exists with GSM in an existing frequency plan and provides link adaptation (modulation and coding are adapted for channel conditions).

Radio Protocol Design

The radio protocol strategy in EDGE is to reuse the protocols of GSM/GPRS whenever possible, thus minimizing the need for new protocol implementation. EDGE enhances both the GSM circuit-switched (HSCSD) and packet-switched (GPRS) mode operation.

EDGE includes one packet-switched and one circuit switched mode, EGPRS and ECSD, respectively.

Enhanced GPRS (EGPRS). The EDGE radio link control (RLC) protocol is somewhat different from the corresponding GPRS protocol. The main changes are related to improvements in link quality control scheme. A *link adaptation* scheme regularly estimates the link quality and subsequently selects the most appropriate modulation and coding scheme for the transmission to maximize the user bit rate. The link adaptation scheme offers mechanisms for choosing the best modulation and coding scheme for the radio link. In GPRS only the coding schemes can be changed between two consecutive link layer control (LLC) frames. In the EGPRS even the modulation can be changed. Different coding and modulation schemes enable adjustment for the
robustness of the transmission according to the environment. Another way to handle link quality variations is *incremental redundancy*. In this scheme, information is fi rst sent with very little coding, yielding a high bit rate if decoding is immediately successful. If decoding is not successful, additional coded bits (redundancy) are sent until decoding succeeds. The more coding that has to be sent, the lower the resulting bit rate and the higher the delay. EGPRS supports combined link adaptation and incremental redundancy schemes. In this case, the initial code rate of the incremental redundancy scheme is based on measurements of the link quality. Benefi ts of this approach are the robustness and high throughput of the incremental redundancy operation in
combination with lower delays and lower memory requirements enabled by the adaptive initial code rate. In EGPRS the different initial code rates are obtained by puncturing a different number of bits from a common convolutional code R _ 1/3. The resulting coding schemes are given in Table 15.1. Incremental redundancy operation is enabled by puncturing a different set of bits each time a block is retransmitted, whereby the code rate is gradually decreased toward 1/3 for every new transmission of the block. The selection of the initial modulation and code rate is based on regular measurements of link quality.

Table 15.1 Channel coding scheme in EDGE (PS transmission).

Coding scheme	Gross bit rate (kbps)	Code rate	Modulation	Radio inter-face rate per time-slot (kbps)	Radio inter-face rate on 8 time-slots (kbps)
CS-1	22.8	0.49	GMSK	11.2	
CS-2	22.8	0.63	GMSK	14.5	
CS-3	22.8	0.73	GMSK	16.7	
CS-4	22.8	1.0	GMSK	22.8	
PCS-1	69.2	0.329	8-PSK	22.8	182.4
PCS-2	69.2	0.496	8-PSK	34.3	274.4
PCS-3	69.2	0.596	8-PSK	41.25	330.0
PCS-4	69.2	0.746	8-PSK	51.60	412.8
PCS-5	69.2	0.829	8-PSK	57.35	458.8
PCS-6	69.2	1.000	8-PSK	69.20	553.6

Actual performance of modulation and coding scheme together with channel characteristics form the basis for link adaptation. Channel characteristics are needed to estimate the effects of a switch to another modulation and coding combination and include an estimated S/I ratio, but also time dispersion and fading characteristics (that affect the effi ciency of interleaving).

EGPRS offers eight additional coding schemes. The EGPRS user has eight modulation and coding schemes available compared to four for GPRS. Besides changes in the physical layer, modifi cations in the protocol structure are also needed. The lower layers of the user data plane designed for GPRS are the physical, RLC MAC, and LLC layers. With EDGE functionality, the LLC layer will not require any modifi cations; however, the RLC/MAC layer has to be modifi ed to accommodate features for effi cient multiplexing and link adaptation procedures to support the basically new physical layer in the EDGE. In the case of GSM, EDGE with the existing GSM radio bands offers wireless multimedia, IP-based applications at the rate of 384 kbps with a bit-rate of 48 kbps per time slot and, under good radio conditions, up to 69.2 kbps per time slot.

Enhanced CSD (ECSD). In this case, the objective is to keep the existing GSM CS data protocols as intact as possible. In order to provide higher data rates, multislot

solutions as in ECSD are provided in EDGE. This has no impact on link or system performance. A data frame is interleaved over 22 frames as in GSM, and three new 8-PSK channel coding schemes are defi ned along with the four already existing for GSM. The radio interface rate varies from 3.6 to 38.8 kbps per time slot (see Table 15.2). Fast introduction of EGPRS/ECSD services is possible by reusing the existing transponder rate adapter unit (TRAU) formats and 16 kbps channel structure several times on the Abis interface. Since data above 14.4 kbps cannot be rate adapted to fi t into one 14.4 kbps TRAU frame, TRAU frames on several 16 kbps.

Table 15.2 Channel coding scheme in EDGE (CS transmission).

Channel name	Code rate	Modulation	Radio interface rate/time-slot (kbps)
TCH/F2.4	0.16	GMSK	3.6
TCH/F4.8	0.26	GMSK	6.0
TCH/F9.6	0.53	GMSK	12.0
TCH/F14.4	0.64	GMSK	14.5
ECSD TCS-1 (NT + T)	0.42	8-PSK	29.0
ECSD TCS-2 (T)	0.46	8-PSK	32.0
ECSD TCS-3 (NT)	0.56	8-PSK	38.8

channels are used to meet the increased capacity requirement. In this case the BTS is required to handle a higher number of 16 kbps Abis channels than time slots used on the radio interface. The benefi t of using the current TRAU formats is that the introduction of new channel coding does not have any impact on the Abis transmission, but it makes it possible to hide the new coding from the TRAU. On the other hand, some additional complexity is introduced in the BTS due to

modifi ed data frame handling. Instead of reusing the current Abis transmission formats for EDGE, new TRAU formats and rate adaptation optimized for increased capacity is specifi ed.

The physical layer can be dimensioned statically for the maximum user rate specified for particular EDGE service or more dynamic reservation of Abis transmission resources

can be applied. The Abis resources can even be released and reserved dynamically during the call, if the link adaptation is applied.The channel coding schemes defi ned for EDGE in PS transmission are listed in Table 15.1 and in CS transmission in Table 15.2.

Services Offered by EDGE

PS Services. The GPRS architecture provides IP connectivity from the mobile station to an external fi xed IP network. For each service, a QoS profi le is defi ned.The QoS parameters include priority, reliability, delay, and maximum and mean bit rate. A specifi ed combination of these parameters defi nes a service, and different services can be selected to suit the needs of different applications.

CS Services. The current GSM standard supports both transparent (T) and nontransparent (NT) services. Eight transparent services are defi ned, offering constant bit rates in the range of 9.6 to 64 kbps.

A nontransparent service uses RLP to ensure virtually error-free data delivery.For this case, there are eight services offering maximum user bit rates from 4.8 to 57.6 kbps. The actual user bit rate may vary according to channel quality and the resulting rate of transmission.The introduction of EDGE implies no change of service defi nitions. The bit rates are the same, but the way services are realized in terms of channel coding is different. For example, a 57.6 kbps nontransparent service can be realized with coding scheme ECSD TCS-1 (telephone control channel-1) and two time slots, while the same service requires four time slots with standard GSM using coding scheme TCH/F14.4. Thus, EDGE CS transmission makes the high bit rate services available with fewer time slots, which is advantageous from a terminal implementation perspective. Additionally, since each user needs fewer time slots, more users can be accepted which increases the capacity of the system.

Asymmetric Services Due to Terminal Implementation. ETSI has standardized two mobile classes: one that requires only GMSK transmission in the uplink and 8-PSK in the downlink and one that requires 8-PSK in both links. For the first class, the uplink bit rate is limited to that of GSM/GPRS, while the EDGE bit rate is still provided in the downlink. Since most services are expected to require higher bit rates in the downlink than in the uplink, this is a way of providing

attractive services with a low complexity mobile station. Similarly, the number of time slots available in the uplink and downlink need not be the same. However, transparent services will be symmetrical.

EDGE Implementation

EDGE makes use of the existing GSM infrastructure in a highly effi cient manner. Radio network planning will not be greatly affected since it will be possible to reuse many existing BTS sites. GPRS packet switching nodes will be unaffected, because they function independently of the user bit rates, and any modifications to the switching nodes will be limited to software upgrades. There is also a smooth evolution path defined for terminals to ensure that EDGE-capable terminals will be small and competitively priced.

EDGE-capable channels will be equally suitable for standard GSM services, and no special EDGE, GPRS, and GSM services will be needed. From an operator viewpoint this allows a seamless introduction of new EDGE services — perhaps starting with the deployment of EDGE in the service hot spots and gradually expanding coverage as demand dictates. The roll-out of EDGE-capable BSS hardware can become part of the ordinary expansion and capacity enhancement of the network. The wideband data capabilities offered by EDGE allows a step-by-step evolution to IMT-2000, probably through a staged deployment of the new 3G air interface on the existing core GSM network. Keeping GSM as the core network for the provision of 3G wireless services has additional commercial benefits. It protects the investment of existing operators; it helps to ensure the widest possible customer base from the outset; and it fosters supplier competition through the continuous evolution of systems.

GSM operators who win licenses in new 2 GHz bands will be able to introduce IMT-2000 wideband coverage in areas where early demand is likely to be greatest. Dual-mode EDGE/IMT-2000 mobile terminals will allow full roaming and handoff from one system to the other, with mapping of services between the two systems. EDGE will contribute to the commercial success of the 3G system in the vital early phases by

ensuring that IMT-2000 subscribers will be able to enjoy roaming and interworking globally.

Building on an existing GSM infrastructure will be relatively fast and inexpensive, compared to establishing a total 3G system. The intermediate move to GPRS and later to EDGE will make the transition to 3G easier. While GPRS and EDGE require new functionality in the GSM network,

with new types of connections to external packet data networks they are essentially extensions of GSM. Moving to a GSM/IMT-2000 core network is likewise a further extension of this network.

Table 15.3 Comparison of GSM data services.

Service type	Data unit	Max. sustained user data rate	Technology	Resources used
Short message service (SMS)	Single 140 octet packet	9 bps	Simplex circuit	SDCCH or SACCH
Circuit-switched data	30 octet frames	9600 bps	Duplex circuits	TCH
HSCSD	192 octet frames	115 kbps	Duplex circuits	1-8 TCH
GPRS	1600 octet frames	171 kbps	Virtual circuit/ packet switching	PDCH (1-8 TCH)
EDGE		384 kbps	Virtual circuit/ packet switching	1-8 TCH

Note: SDCCH: Stand-alone Dedicated Control Channel; SACCH: Slow Associated Control Channel; TCH: Traffic Channel; PDCH: Packet Data Channel (all refer to GSM logical channels).

EDGE provides GSM operators — whether or not they get a new 3G license — with a commercially attractive solution to develop the market for wideband multimedia services. Familiar interfaces such as the Internet, volume-based charging, and a progressive increase in available user data rates will remove some of the barriers to large-scale take-up of wireless data services. The way forward to 3G services will be a staged evolution from today's GSM data services through GPRS and EDGE.

Increased user data rates over the radio interface requires redesign of physical transmission methods, frame formats, and signaling protocols in different network

interfaces. The extent of the modification needed depends on the user data rate requirement, i.e., whether the support of higher data is required or merely a more effi cient use of the radio time slot to support current data services is needed.

Several alternatives to cover the increased radio interface data rates on the Abis interface for EGPRS and ECSD can be envisioned. The existing physical structure can be reused as much as possible or a new transmission method optimized for EDGE can be specifi ed. Table 15.3 provides a comparison of GSM data services with GPRS and EDGE.

5.Explain in detail about Third-Generation (3G) Wireless Systems. (L-2,CO-4)

The International Telecommunication Union (ITU) began studies on the globalization of personal communications in 1986 and identifi ed the long-term spectrum requirements for the future *third-generation* (*3G*) mobile wireless telecommunications systems. In 1992, the ITU identifi ed 230 MHz of spectrum in the 2 GHz band to implement the IMT-2000 system on a worldwide basis for satellite and terrestrial components. The aim of IMT-2000 is to provide universal coverage enabling terminals to have seamless roaming across multiple networks. The ITU accepted the overall standardization responsibility of IMT-2000 to defi ne radio interfaces that are applicable in different radio environments including indoor, outdoor, terrestrial, and satellite [23–28, 30–32]. Figure 15.9 provides an overview of the IMT family. *IMT-DS* is the direct spread (DS) technology and includes WCDMA systems. This technology is intended for UMTS terrestrial radio access (UTRA)-FDD and is used in Europe and Japan. *IMT-TC* family members are the UTRA-TDD system that uses timedivision (TD) CDMA, and the Chinese TD-synchronous CDMA (TD-SCDMA). Both standards are combined and the third-generation partnership project (3GPP) is responsible for the development of the technology. *IMT-MC* includes multiple carrier (MC) cdma2000 technology, an evolution of the cdmaOne family. 3GPP2 is responsible for standardization. *IMT-SC* is the enhancement of the US TDMA systems. UWC-136 is a single carrier (SC) technology. This technology applies EDGE to enhance the 2G IS-136 standard. It is now integrated into the 3GPP efforts. *IMT-FT* is a frequency time (FT) technology. An enhanced version of the cordless telephone standard digital European cordless technology

(DECT) has

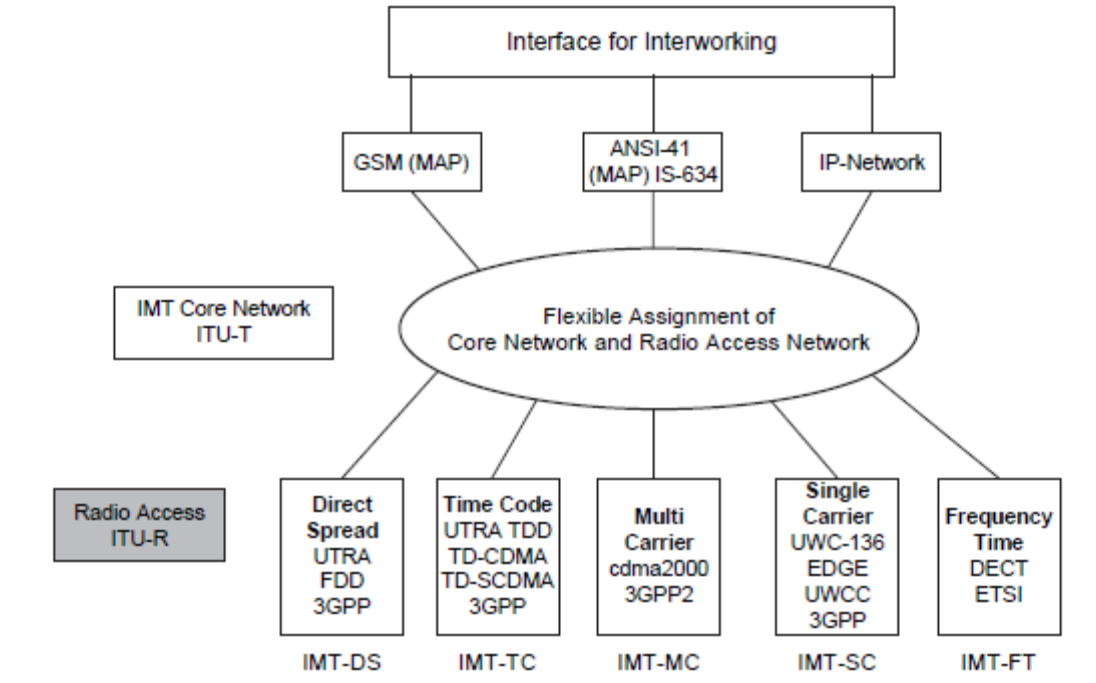


Figure 15.9 IMT family.

been selected for low mobility applications. The ETSI has the responsibility for standardization of DECT. 3G mobile telecommunications systems are intended to provide worldwide access and global roaming for a wide range of services. Standards bodies in Europe, Japan, and North America are trying to achieve harmony on key and interrelated issues including radio interfaces, system evolution and backward compatibility, user's migration and global roaming, and phased introduction of mobile services and capabilities to support terminal mobility. Universal Mobile Telecommunication System (UMTS) studies were carried out by ETSI in parallel with IMT-2000

to harmonize its efforts with ITU. In Japan and North America, standardization efforts for 3G were carried out by the Association of Radio Industries Business (ARIB) and the TIA committee TR45, respectively. Two partnership projects, 3GPP and 3GPP2, are involved in harmonizing 3G efforts in Europe, Japan, and North America.In Europe, 3G

systems are intended to support a substantially wider and enhanced range of services compared to the 2G (GSM) system. These enhancements include multimedia services, access to the Internet, high rate data, and so on. The enhanced services impose additional requirements on the fi xed network functions to support mobility. These requirements are achieved through an evolution path to capitalize on the investments for the 2G system in Europe, Japan, and North America. In North America, the 3G wireless telecommunication system, cdma2000 was proposed to ITU to meet most of the IMT requirements in the indoor offi ce, indoor to outdoor pedestrian, and vehicular environment. In addition, the cdma2000 satisfi es the requirements for 3G evolution of 2G TIA/EIA 95 family

of standards (cdmaOne). In Japan, evolution of the GSM platform is planned for the IMT (3G) core network due to its fl exibility and widespread use around the world. Smooth migration from GSM to IMT-2000 is possible. The service area of the 3G system overlays with the existing 2G (PDC) system. The 3G system connects and interworks with 2G systems through an interworking function (IWF). An IMT-2000-PDC dual mode terminal as well as the IMT-2000 single mode terminal are deployed. UMTS as discussed today and introduced in many countries is based on the initial release of UMTS standards referred to as release 99 or R99. This (release)

is aimed at a cost-effective migration from GSM to UMTS. After R99 the release of 2000 or R00 followed. 3GPP decided to split R00 into two standards and call them release 4 (Rel-4) and release 5 (Rel-5). The version of all standards fi nalized for R99 is now referred to as Rel-3 by 3GPP. Rel-4 introduces QoS in the fi xed network plus several execution environments (e.g., MExE, mobile execution environment) and new service architectures. Rel-4 was suspended in March 2001. Rel-5 specifi es a new core network. The GSM/GPRS-based core network will be

replaced by an almost all-IP core network. The content of Rel-5 was suspended in March 2002. This standard integrates IP-based multimedia services (IMS) controlled by the IETF’s session initiation protocol (SIP). A high-speed downlink packet access (HSDPA) service with 8 to 10 Mbps was included. Also, a wideband16 kHz adaptive multirate (AMR) codec was added for better quality. End-toend QoS and several data compression techniques were added. 3GPP is currently working on Rel-6. This

standard includes multiple input multiple output (MIMO) antennas, enhanced multimedia service (MMS), security enhancement, WLAN/WWAN interworking, broadcast/multicast services, enhanced IMS, IP emergency call, and many more management features. A primary assumption for UMTS is that it is based on an evolved GSM core network. This provides backward compatibility with GSM in terms of network protocols and interfaces (MAP, ISUP (ISDN user part), etc.) The core network supports both GSM and UMTS/IMT-2000 services, including handoff and roaming between the two (see Figure 15.10). The proposed W-CDMA based UMTS terrestrial radio access network (UTRAN) is connected to the GSM-UMTS core network using a new multi-vendor interface (*Iu*). The transport protocol within the new radio network and the core network will be IP. There is a clear separation between the services provided by UTRAN and the actual channels used to carry these services. All radio network functions (such as resource control) are handled within the radio access network and clearly separated from the service and subscription functions in the UMTS core network (UCN). The GSM-UMTS network, shown in Figure 15.11, consists of three main
entities:

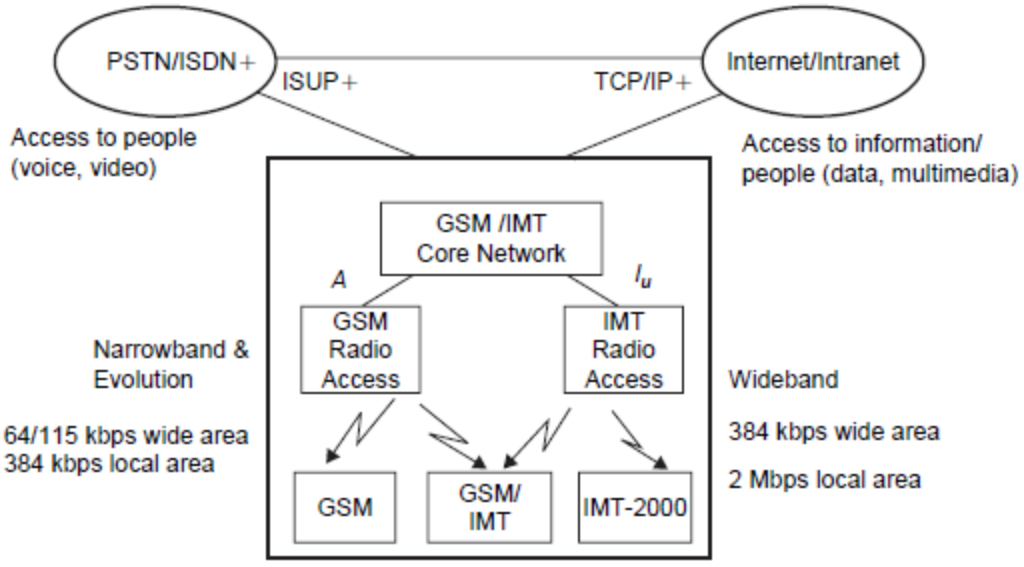


Figure 15.10 Evolution to UMTS/IMT-2000 in a GSM environment.

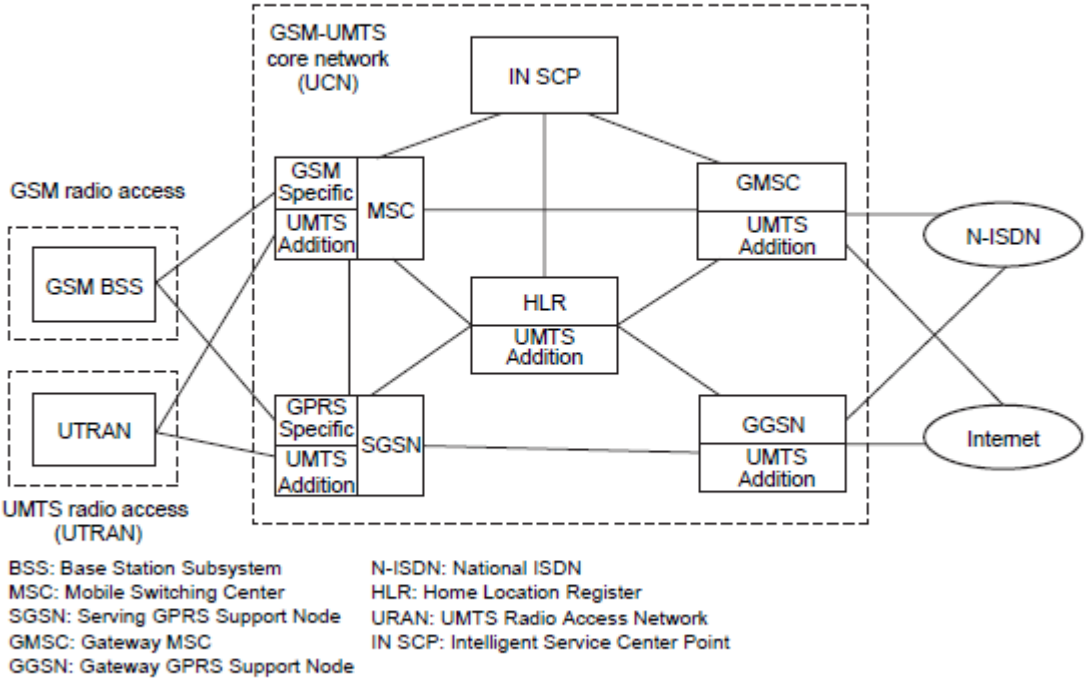


Figure 15.11 General GSM-UMTS network architecture.

GSM-UMTS core network (UCN)

UMTS terrestrial radio access network (UTRAN)

GSM base station subsystem (BSS)

Like the GSM-GPRS core network, the GSM-UMTS core network has two different parts: a circuit-switched MSC and a packet-switched GRPS support node (GSN). The core network access point for GSM circuit-switched connections is the GSM MSC, and for packet-switched connections it is the SGSN. GSM-defi ned services (up to and including GSM Phase 2_) are supported in the usual GSM manner. The GSM-UMTS core network implements supplementary

services according to GSM principles (HLR-MSC/VLR). New services, beyond Phase 2_ are created using new service capabilities. These service capabilities may be seen as building blocks for application development and include:

Bearers defined by QoS

Mobile station execution environment (MExE)
Telephony value-added services (TeleVAS)
Subscriber identity module (SIM) toolkit
Location services
Open interfaces to mobile network functions
Down-loadable application software
Intelligent Network/Customized Applications for Mobile Enhanced Logic(IN/CAMEL) and service nodes. In addition to new services provided by the GSM-UMTS network itself, many new services and applications will be realized using a client/server approach, with servers residing on service local area networks (LANs) outside the GSM-UMTS core network. For such services, the core network simply acts as a transparent bearer. This approach is in line with current standardization activities, and is important from a service continuity point of view. The core network will ultimately be used for the transfer of data between the end-points, the client and the server.

IN techniques are one way to provide seamless interworking across GSMUMTS networks. CAMEL already provides the basis for GSM/IN interworking. The IN infrastructure may be shared by fi xed and mobile networks, and can support fi xed/mobile service integration, as needed by International Mobile Telecommunications (IMT). The inherent support for third-party service providers in IN means such providers could offer all or part of the integrated services. This role of IN is already apparent in services such as virtual private networks (VPNs), regional

subscription and one number, which are available as network-independent and customer-driven services. Service nodes and IN can play a complementary role. IN is suitable for subscription

control and group services where high service penetration in a very wide area with frequent service invocation are more important than sophistication. Service nodes are better for providing differentiated user interfaces; e.g., personalcall and messaging services that use advanced in-band processing and span several access networks.

To make the most of the new radio access network's capabilities, and to cater to the large increase in data traffic volume, asynchronous transmission mode (ATM) is used as the transport protocol within UTRAN and toward the SM-UMTS core network. The

combination of an ATM cell-based transport network, WCDMA's use of variable-rate speech coding with improved channel coding, and an increased volume of packet data traffi c over the air interface will mean a saving of about 50% in transmission costs, compared with equivalent current solutions. ATM, with the newly standardized AAL2 adaptation layer, provides an effi cient transport protocol, optimized for delay-sensitive speech services and packet data services. Statistical multiplexing in ATM provides maximum utilization of existing and new transmission infrastructure throughout the entire network.

In the complex multiservice, multivendor, multiprovider environment of 3G wireless services network management is a critical issue. The growth of packet data traffi c requires new ways of charging for services and new billing systems to support them. There will continue to be a growing demand for better customer care and cost reductions in managing mobile networks, driven by the need to:

Provide sophisticated personal communications services

Expand the customer base beyond the business user base

Separate the service provider and network operator roles

Provide "one-stop" billing for a range of services

New operations and management functions are needed to support new services and network functionality. Standardization of interfaces is critical especially for alignment with current management interfaces in the GSM-UMTS core network. Management information needs to be part of standard traffi c interfaces. With the right service strategy and network planning, GSM operators will be able to capitalize on the wideband multimedia market through a staged evolution of their core networks, with the addition of new radio access technology as it becomes available.

7.Explain the techniques about UMTS Network Reference Architecture. (L-3,CO-4)

A UMTS system can be divided into a set of domains and the reference points that interconnect

them. Figures 15.12 and 15.13 show these domains and reference points. A simplifi ed mapping of functional entities to the domain model is shown in Figure 15.14. Note that this is a reference model and does not represent any

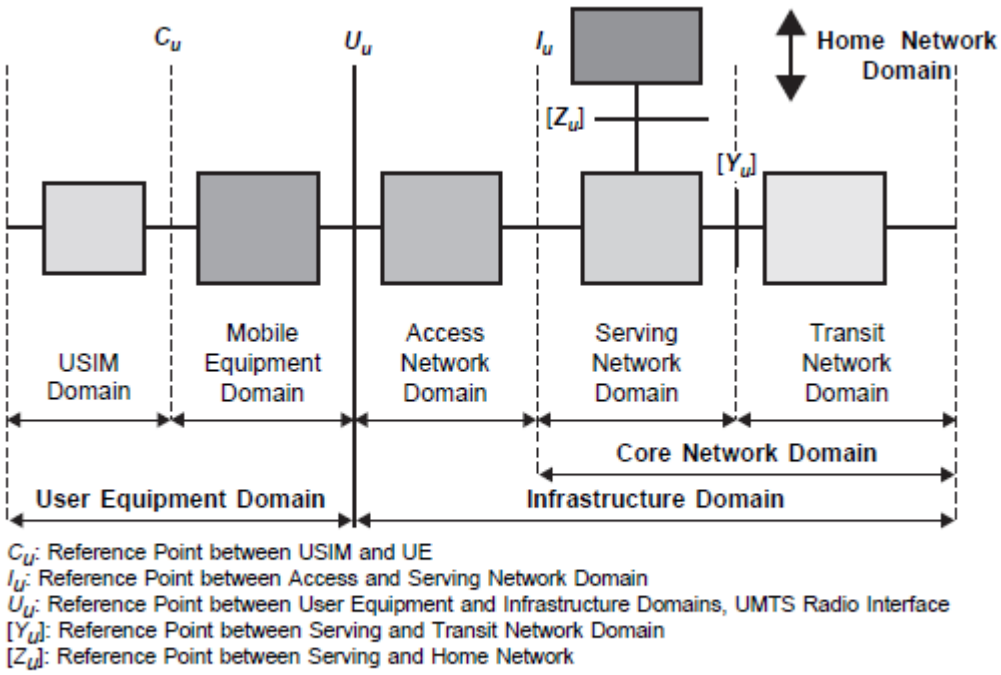


Figure 15.12 UMTS domains and reference points.

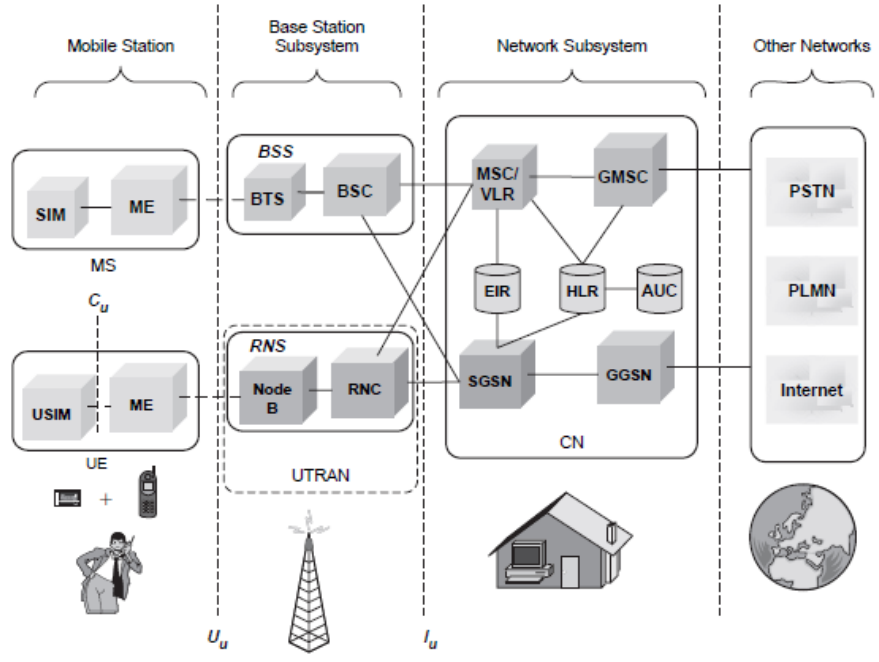


Figure 15.13 UMTS—3G reference architecture.

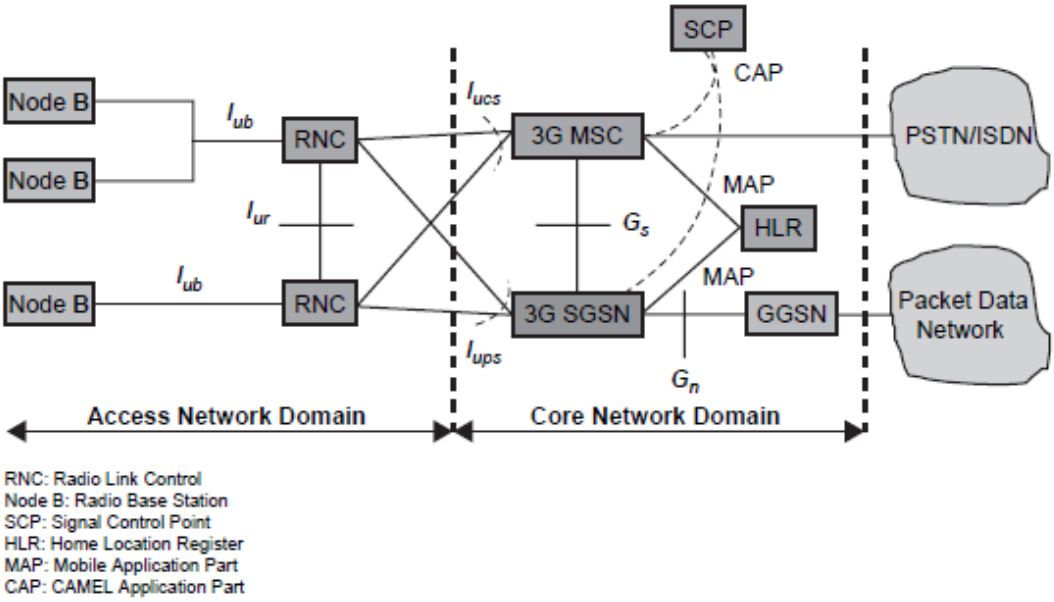


Figure 15.14 Simplified UMTS network reference model.

architecture. The *Iu* is split functionally into two logical interfaces, *Iups* connecting the packet switched domain to the access network and the *Iucs* connecting the circuit switched domain to the access network. The standards do not dictate that these are physically separate, but the user plane for each is different and the control plane may be different. The *Iur* logically connects radio network controllers (RNCs) but could be physically realized by a direct connection between RNCs or via the core network.

15.5 Channel Structure in UMTS Terrestrial Radio(H-1,CO-4)

Access Network

The UMTS terrestrial radio access network (UTRAN) [10–22] has an *access stratum* and *nonaccess stratum*. The access stratum includes air interface and provides functions related to OSI layer 1, layer 2, and the lower part of layer 3. The non-access stratum deals with communication between user equipment (UE) and core network (CN) and includes OSI layer 3 (upper part) to layer 7. The radio interface, *Uu*, is the interface between UE and UTRAN. It consists of three protocol layers: physical layer, data link layer, and network layer (see Figure 15.15). The radio interface provides physical channels to carry data over the radio path and logical channels to carry a particular type of data. There are two types of logical channels: signaling and control, and traffi c

channel. The *physical layer* in UTRAN performs the following functions: Forward error correction, bit-interleaving, and rate matching Signal measurements,
Micro-diversity distribution/combining and soft handoff execution, Multiplexing/mapping of services on dedicated physical codes

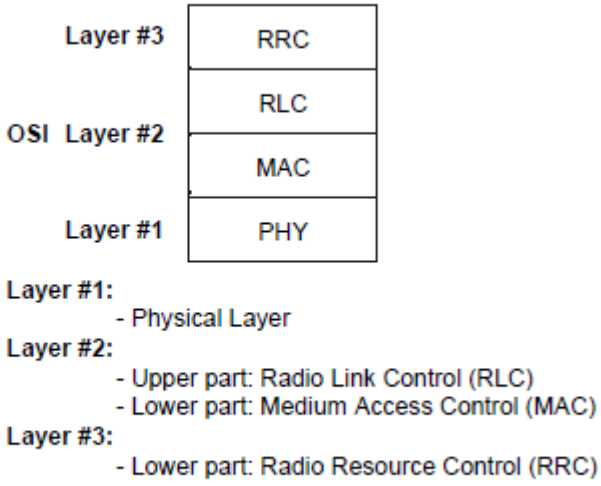


Figure 15.15 OSI layer model and air interface protocols.

Modulation, spreading, demodulation, despreading of physical channels
Frequency and time (chip, bit, slot, frame) synchronization
Fast closed-loop power control
Power weighting and combining of physical channels
Radio frequency (RF) processing.
The *medium access control sublayer* is responsible for efficiently transferring data for both real-time (CS) and non-real-time (PS) services to the physical layer. MAC offers services to the radio link control (RLC) sublayer and higher layers. The MAC layer provides data transfer services on logical channels. MAC is responsible for
Selection of appropriate transport format (basically bit rate) within a predefi ned set, per information unit delivered to the physical layer
Service multiplexing on *random access channel* (*RACH*), *forward access channel* (*FACH*), and *dedicated channel* (*DCH*)
Priority handling between data fl ow of a user as well as between data flows from several users

Access control on RACH and FACH

Contention resolution on RACH

Radio link control (*RLC*) sets up a logical link over the radio interface and is responsible for fulfilling QoS requirements. RLC responsibilities include:

Segmentation and assembly of the packet data unit

Transfer of user data

Error correction through retransmission

Sequence integrity

Duplication information detection

Flow control of data

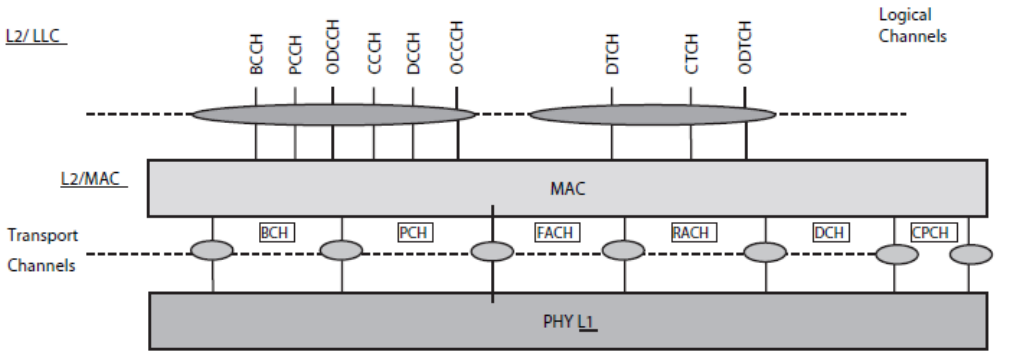
The *radio resource control* (*RRC*) layer broadcasts system information, handles radio resources (i.e., code allocation, handover, admission control, and measurement/control report), and controls the requested QoS. The RRC layer offers the following services to the core network:

General control (GC) service used as an information broadcast service

Notification (Nt) service used for paging and notification of a selected UE

Dedicated control (DC) service used to establish/release a connections and transfer messages

The channels in UTRAN are physical, transport, and logical (see Figure 15.16).



• Physical channels: identified by code, frequency, phase (*I*/*Q*) (UL only) and time slot (TDD only)

• Transport channels: offer information transfer services to the MAC layer

• Logical channels: are used by MAC layer to provide data transport services

BCCH: Broadcast Control Channel
PCCH: Paging Control Channel
DCCH: Dedicated Control Channel
ODCCH: ODMA Dedicated Control Channel
OCCCH: ODMA Common Control Channel
DTCH: Dedicated Traffic Channel
ODTCH: ODMA Dedicated Traffic Channel
CTCH: Common Traffic Channel
CCCH: Common Control Channel

Figure 15.16 UTRAN channels.

The functions of *logical control channels* and *logical traffi c channels* in UTRAN are listed in Tables 15.4 and 15.5.

In UTRAN *transport channels* can be either common (i.e., shared between users) or dedicated channels. They offer information transfer services to the MAC sublayer. Dedicated transport channels are *dedicated channel* (*DCH*) (*up link*, *UL* and *down link*, *DL*), *fast uplink signaling channel* (*FAUSCH*), and ODMA

Table 15.4 Logical control channel in UTRAN.

Channel	Function
Broadcast control channel (BCCH)	DL channel for broadcasting system and control information
Paging control channel (PCCH)	DL channel to transfer page information, used when: (1) network does not know the location of cell of the mobile, and (2) mobile is in cell connected state (using sleep mode)
Common control channel (CCCH)	Bidirectional channel to transfer control information between network and mobile, it is used: (1) by mobile without RRC connection with the network, and (2) by mobile using common transport channel to access a new cell after cell resection
Dedicated control channel (DCCH)	Point-to-point bidirectional channel to transmit dedicated information between a mobile and network. The channel is established through RRC connection setup procedure
ODMA common control channel (OCCCH)	Bidirectional channel to transmit control information between mobiles
ODMA dedicated control channel (ODCCH)	Point-to-multipoint bidirectional channel to transmit dedicated control information between mobiles. This channel is established through RRC connection setup procedure

Table 15.5 Logical traffic channels in UTRAN.

Channel	Function
Dedicated traffic channel (DTCH)	Point-to-point, dedicated to one mobile to transfer user information. A DTCH can exist in both UL and DL.
ODMA traffic channel (ODTCH)	Point-to-point channel dedicated to one mobile to transfer user information between mobiles. An ODTCH can exist in relay link. A point-to-multipoint unidirectional channel to transfer dedicated user information for all or a group of specified mobiles.

(*opportunity driven multiple access*) *dedicated channel* (*ODCH*). The common DL transport channels are listed in Table 15.6.

The mapping between logical and transport channels is given below (see Figure 15.17):

BCCH is connected to BCH

PCCH is connected to PCH

CCCH is connected to RACH and FACH

Table 15.6 UTRAN common transport channels.

Channel	Function
Broadcast channel (BCH)	DL channel used to broadcast system- and cell-specific information, transmitted over the entire cell with low fixed bit rate
Forward access channel (FACH)	DL channel transmitted over the entire or only a part of cell using beam-forming antennas, uses slow power control
Paging channel (PCH)	DL channel transmitted over the entire cell, transmission of PCH is associated with the transmission of a physical layer signal, the paging indicator, to support efficient sleep mode procedure
Random access channel (RACH)	UL channel received over the entire cell, characterized by a limited size data field, a collision risk, and by use of open loop power control
Common packet channel (CPCH)	UL channel, contention-based random access channel used for transmission of bursty data traffic, associated with a DCH on DL, which provides power control for the UL CPCH
Downlink shared channel (DSCH)	DL channel shared by several mobiles, associated with a DCH

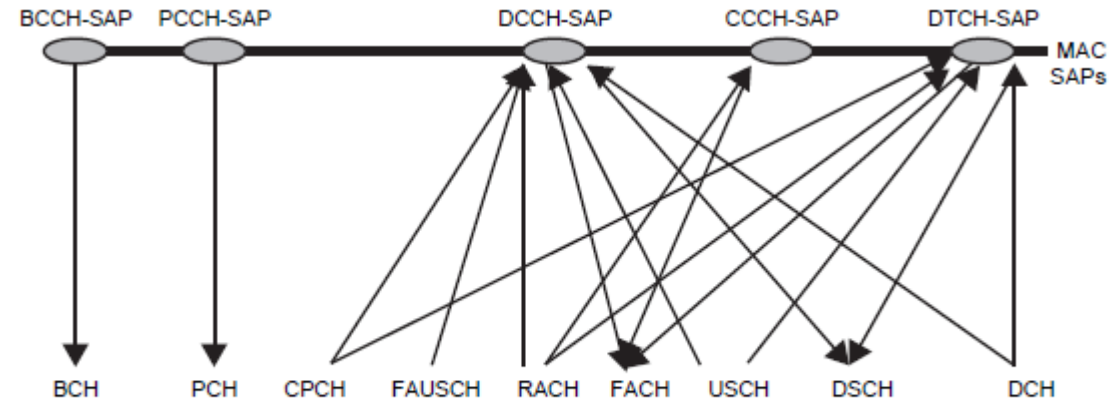


Figure 15.17 Mapping between logical and transport channels in UTRAN.

DTCH can be connected to either RACH and FACH and DSCH, to DCH and DCH, to a DCH, to a CPCH (FDD mode only)

CTCH can be connected to DSCH, FACH, or BCH

DCCH can be connected to either RACH and FACH, to RACH and DSCH,
to DCH and DSCH, to a DCH, a CPCH (TDD mode only), to FAUSCH,CPCH (FDD mode only).

In UTRAN, the basic physical resource is a *physical channel* identifi ed by code and frequency. Physical channels consist of radio frames and time slots (see Figure 15.18). The length of a radio frame is 10 ms and one frame consists of 15 time slots. For DL channels two codes are used, one to identify the cell and the other to identify a particular channel within that cell. For UL a long code is used to identify the channel. The UL channel uses different data streams transmitted on
the I and Q branch. A physical channel corresponds to a specifi c carrier frequency,code(s), and on UL a relative phase (0, _/2). The UL dedicated physical channel (DPCH) is a user dedicated, point-topoint channel between UE and node B. These channels carry dedicated channels at various rates up to 2 Mbps. The UL-dedicated physical data channels are I/Q (i.e., DPDCH on I-branch and DPCCH on Q-branch) and code multiplexed. There are two types of DPCH: (1) *dedicated physical data channel* (*DPDCH*) to carry user data and signaling information generated at layer 2 (there may be none, one, or several DPDCHs); and (2) *dedicated physical control channel* (*DPCCH*) to carry control information generated at layer 1 (pilot bits, transmit power control (TPC) commands, feedback information (FBI) commands, and optional transport
format combination indicator (TFCI)). For each layer 1 connection, there is only one UL DPCCH. DPCCH rate and power remain constant. The UL dedicated physical channel carries 10 _ 2*k* (*k* _ 0, 1, . . . , 6) bits per slot and may have a spreading factor (SF) from 256 and 4.

The UL common physical channels are *physical random access channel* (*PRACH*) used to carry the RACH and fast uplink signaling channel (FAUSCH) and *physical common packet channel* (*PCPCH*) to carry CPCH. The DL dedicated physical data channel is time multiplexed. The DL dedicated physical channel carries 10 _ 2*k* (*k* _ 0, 1, . . . , 7) bits per slot and may have spreading factor (SF _ 512/2*k*) from 512 to 4 (see Figure 15.18). The DL common physical channels include the following channels: *Primary common control physical channel* (*PCCPCH*) carries BCH, rate 30 kbps, SF _ 256;

continuous transmission; no power control *Secondary common control physical channel* (*SCCPCH*) carries FACH and PCH, transmitted when data is available; SF range is from 256 to 4 *Synchronization channel* (*SCH*) is used for cell search. It consists of two sub-channels: primary SCH transmits a modulated code of 256 chips once every slot, and secondary SCH transmits repeatedly 15 codes of 256 chips.

15.5 Channel Structure in UMTS Terrestrial Radio Access Network 503

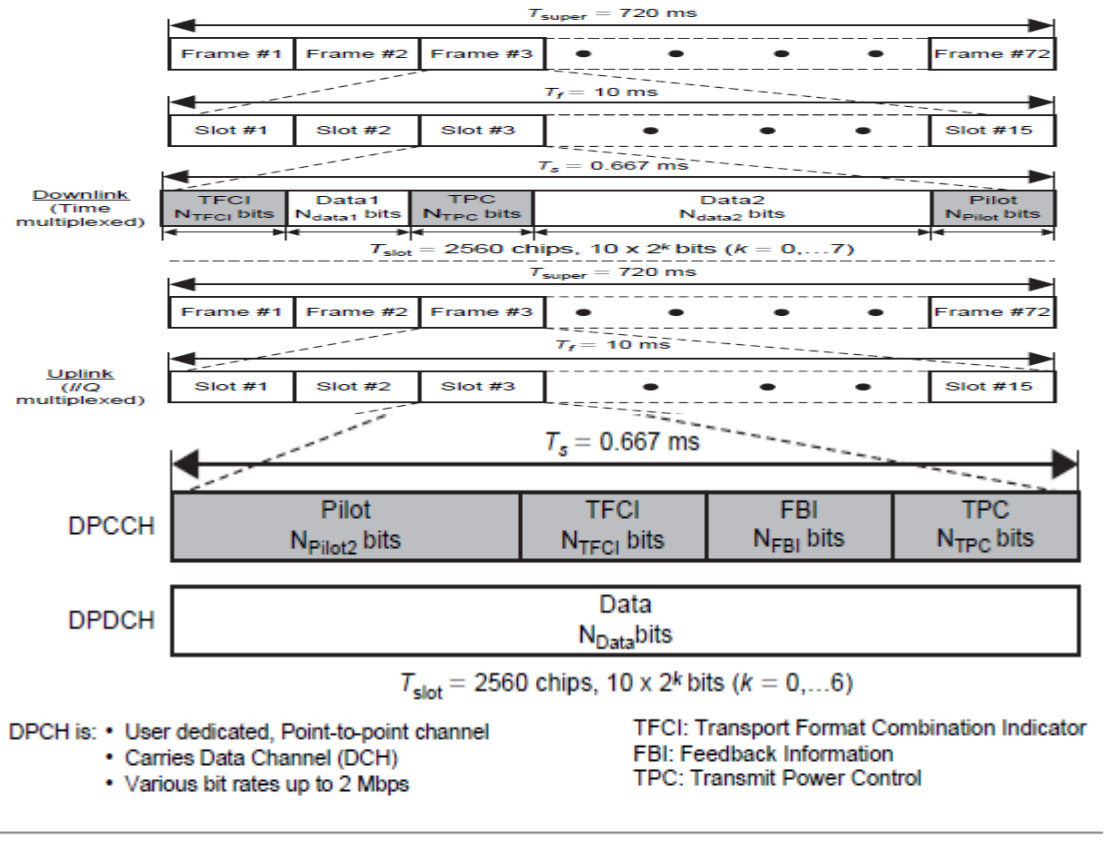


Figure 15.18 UTRA dedicated physical channels (DPCHs) framing.

Physical downlink shared channel (*PDSCH*) carries DSCH; shared by users based on code multiplexing; associated with DPCH. *Acquisition indicator channel* (*AICH*) carries acquisition indicators. *Page indicator channel* (*PICH*) carries a page for UE; fi xed rate, SF _ 256

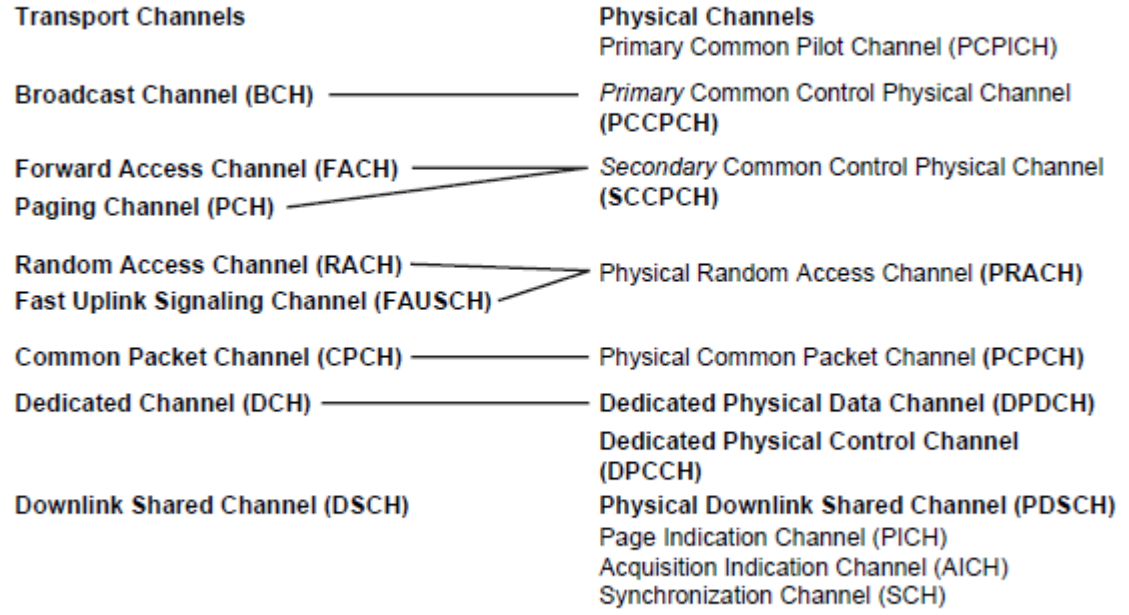


Figure 15.19 Mapping between transport and physical channels in UTRA.

Primary common pilot channel (*PCPICH*) is used as phase reference for SCH, PCCPCH, AICH, PICH, and default phase reference for all other DL physical channels, one per cell and broadcast over entire cell. *Secondary common pilot channel* (*SCPICH*) is a continuous channel with the same spreading and scrambling codes, transmitted on different antennas in case of DL transmit diversity; SF _ 256. It may be transmitted over only part of the cell.

The mapping between transport and physical channels is shown in Figure 15.19.

9.How the Spreading and Scrambling techniques used in UMTS. (L-1,CO-4)

In a WCDMA system, isolation between users in the downlink is accomplished through the combination of user-specifi c channelization codes and cell-specific scrambling codes. Channelization codes are generated recursively to form a binary tree structure

(see Figure 15.20). Spreading is used in combination with scrambling. Scrambling is used on top of spreading to separate mobile terminals or cells from each other. Scrambling does not change the chip rate nor the bandwidth. Transmissions from a single source are separated by channelization codes based on an orthogonal variable spreading factor (OVSF) technique (see Appendix D) to allow spreading to be changed while maintaining orthogonality between codes.When a code is intended to be used, no other code generated from the intended code can be used; no code between the intended code and the root code can be used. These restrictions apply to individual sources; they do not apply to different cells in downlink or different mobile stations in the uplink. Table 15.7 lists the characteristics of synchronization, channelization, and scrambling codes used in a WCDMA system.

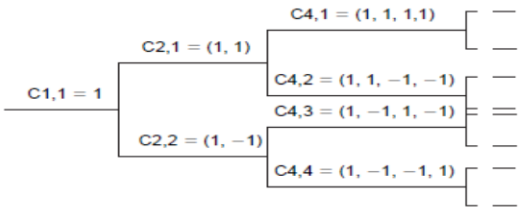


Figure 15.20 Channelization code tree.

Table 15.7 Synchronization, channelization, and scrambling codes of a WCDMA system.

	Synchronization codes	Channelization codes	Scrambling codes, uplink	Scrambling codes, downlink
Type	Gold codes, primary synchronization code (PSC) and secondary synchronization code (SSC)	OVSF, sometimes called Walsh codes	Complex-valued gold code segments (long) or complex-valued S(2) codes (short)	Complex-valued gold code segments. Pseudo noise (PN) codes
Length	256 chips	4–512 chips	38,400 chips/256 chips	38,400 chips
Duration	66.67 μs	1.04 μs–133.34 μs	10 ms/66.67 μs	10 ms
Number of codes	1 primary code/16 secondary codes	Spreading factor: UL: 4–256 DL: 4–512	16,777,216	512 primary/ 15 secondary for each primary code
Spreading	No, does not change bandwidth	Yes, increases bandwidth	No, does not change bandwidth	No, does not change bandwidth
Usage	To enable terminal to locate and synchronize to the cells' main control channels	UL: To separate physical data and control data from same terminal DL: To separate connection to different terminals in a cell	Separation of terminals	Separation of cells

15.7 UMTS Terrestrial Radio Access Network Overview

The UTRAN consists of a set of radio network subsystems (RNSs) (see Figure 15.21). The RNS has two main logical elements: Node B and an RNC. The RNS is responsible for the radio resources and transmission/reception in a set of cells. A cell (sector) is one coverage area served by a broadcast channel. An RNC is responsible for the use and

allocation of all the radio resources of the RNS to which it belongs. The RNC also handles the user voice and packet data traffi c, performing the actions on the user data streams that are necessary to access the radio bearers. The responsibilities of an RNC are:

Intra UTRAN handover

Macro diversity combining/splitting of *Iub* data streams

Frame synchronization

Radio resource management

Outer loop power control

Iu interface user plane setup

Serving RNS (SRNS) relocation

Radio resource allocation (allocation of codes, etc.)

Frame selection/distribution function necessary for soft handover (functions

of UMTS radio interface physical layer)

UMTS radio link control (RLC) sublayers function execution

Termination of MAC, RLC, and RRC protocols for transport channels,

i.e., DCH, DSCH, RACH, FACH

Iub's user plane protocols termination

A Node B is responsible for radio transmission and reception in one or more cells to/from the user equipment (UE). The logical architecture for Node B is

shown in Figure 15.22.

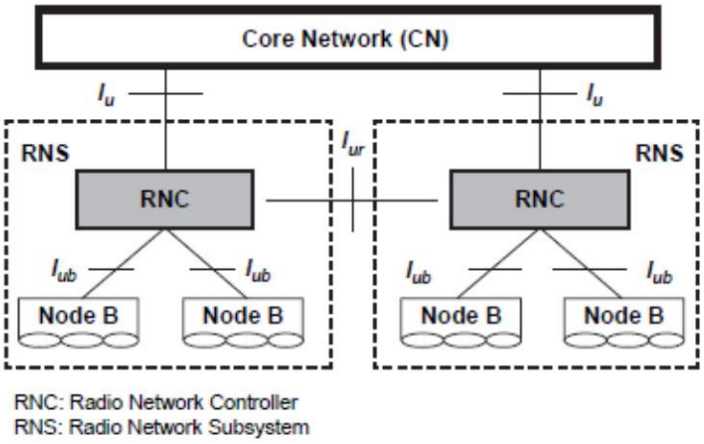


Figure 15.21 UTRAN logical architecture.

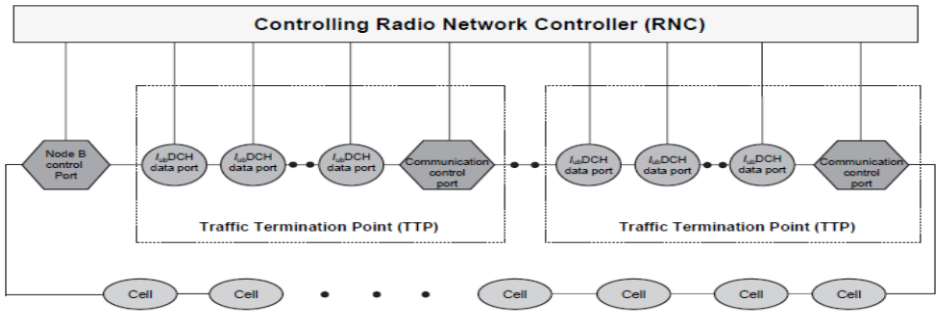


Figure 15.22 Node B logical architecture.

The following are the responsibilities of the Node B:

Termination of *Iub* interface from RNC

Termination of MAC protocol for transport channels RACH, FACH

Termination of MAC, RLC, and RRC protocols for transport channels:

BCH, PCH

Radio environment survey (BER estimate, receiving signal strength, etc.)

Inner loop power control

Open loop power control

Radio channel coding/decoding

Macro diversity combining/splitting of data streams from its cells (sectors)

Termination of *Uu* interface from UE

Error detection on transport channels and indication to higher layers

FEC encoding/decoding and interleaving/deinterleaving of transport channels

Multiplexing of transport channels and demultiplexing of coded composite transport channels

Power weighting and combining of physical channels

Modulation and spreading/demodulation and despreading of physical channels

Frequency and time (chip, bit, slot, frame) synchronization

RF processing

10.Briefly give the significance about UTRAN Logical Interfaces. (L-2,CO-4)

In UTRAN protocol structure is designed so that layers and planes are logically independent of each other and, if required, parts of protocol structure can be changed in the future without affecting other parts. The protocol structure contains two main layers, the radio network layer

(RNL) and the transport network layer (TNL). In the RNL, all UTRAN-related functions are visible, whereas the TNL deals with transport technology selected to be used for UTRAN but without any UTRAN-specifi c changes. A general protocol model for UTRAN interfaces is shown in Figure 15.23. The control plane is used for all UMTS-specifi c control signaling. It includes the application protocol (i.e., radio access network application part (RANAP) in *Iu*, radio network subsystem application part (RNSAP) in *Iur* and node B application part (NBAP) in *Iub*). The application protocol is used for setting up bearers to the UE. In the three-plane structure the bearer parameters in the application protocol are not directly related to the user plane technology, but rather they are general bearer parameters.

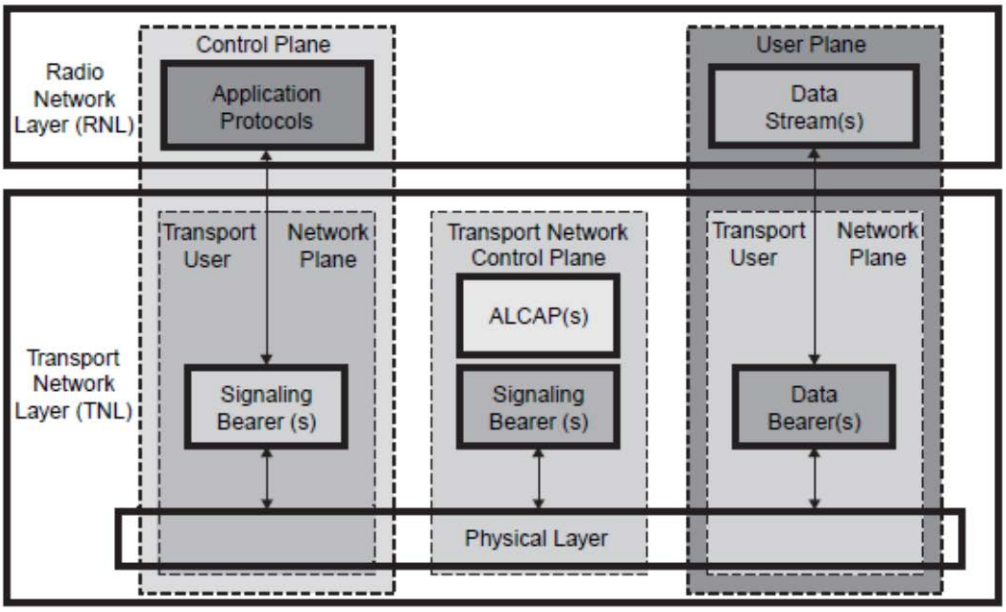


Figure 15.23 General protocol model for UTRAN interfaces.

User information is carried by the user plane. The user plane includes data stream(s), and data bearer(s) for data stream(s). Each data stream is characterized by one or more frame protocols specifi ed for that interface. The transport network control plane carries all control signaling within the transport layer. It does not include radio network layer information. It contains

access link control application part (*ALCAP*) required to set up the transport bearers (data bearers) for the user plane. It also includes the signaling bearer needed for the ALCAP. The transport plane lies between the control plane and the user plane. The addition of the transport plane in UTRAN allows the application protocol in the radio network control plane to be totally independent of the technology selected for the data bearer in the user plane. With the transport network control plane, the transport bearers for data bearers in the user plane are set up in the following way. There is a signaling transaction by application protocol in the control plane that initiates set-up of the data bearer by the ALCAP protocol specifi c for the user plane technology.

The independence of the control plane and user plane assumes that an ALCAP signaling occurs. The ALCAP may not be used for all types of data bearers. If there is

no ALCAP signaling transaction, the transport network control plane is not required. This situation occurs when preconfi gured data bearers are used. Also, the ALCAP protocols in the transport network control plane are not used to set up the signaling bearer for the application protocol or the ALCAP during real-time operation.

_Iu_ Interface

The UMTS *Iu* interface is the open logical interface that interconnects one UTRAN to the UMTS core network (UCN). On the UTRAN side the *Iu* interface is terminated at the RNC, and at the UCN side it is terminated at U-MSC. The *Iu* interface consists of three different protocol planes — the *radio network control plane* (*RNCP*), the *transport network control plane* (*TNCP*), and the *user plane* (*UP*).

The RNCP performs the following functions:

It carries information for the general control of UTRAN radio network operations.

It carries information for control of UTRAN in the context of each specifi c call.

It carries user call control (CC) and mobility management (MM) signaling messages.

The control plane serves two service domains in the core network, the packet-switched (PS) domain and circuit-switched (CS) domain. The CS domain supports circuit-switched services. Some examples of CS services are voice and fax. The CS domain can also provide intelligent services such as voice mail and free phone. The CS domain connects to PSTN/ISDN networks.

The CS domain is expected to evolve from the existing 2G GSM
PLMN.

The PS domain deals with PS services. Some examples of PS services are Internet access and multimedia services. Since Internet connectivity is provided, all services currently available on the Internet such as search engines and e-mail are available to mobile users. The PS domain connects to IP networks. The PS domain is expected to evolve from the GPRS PLMN.

The *Iu* circuit-switched and packet-switched protocol architecture are shown
in Figures 15.24 and 15.25.

The control plane protocol stack consists of RANAP on the top of signaling system 7 (SS7) protocols. The protocol layers are the *signaling connection control part* (*SCCP*), the *message transfer part* (*MTP3-B*), and *signaling asynchronous transfer mode (ATM)*

adaptation layer for network-to-network, interface (SAAL-NNI). The SAAL-NNI is divided into *service-specifi c coordination*

function (*SSCF*), the *service-specifi c connection-oriented protocol* (*SSCOP*),

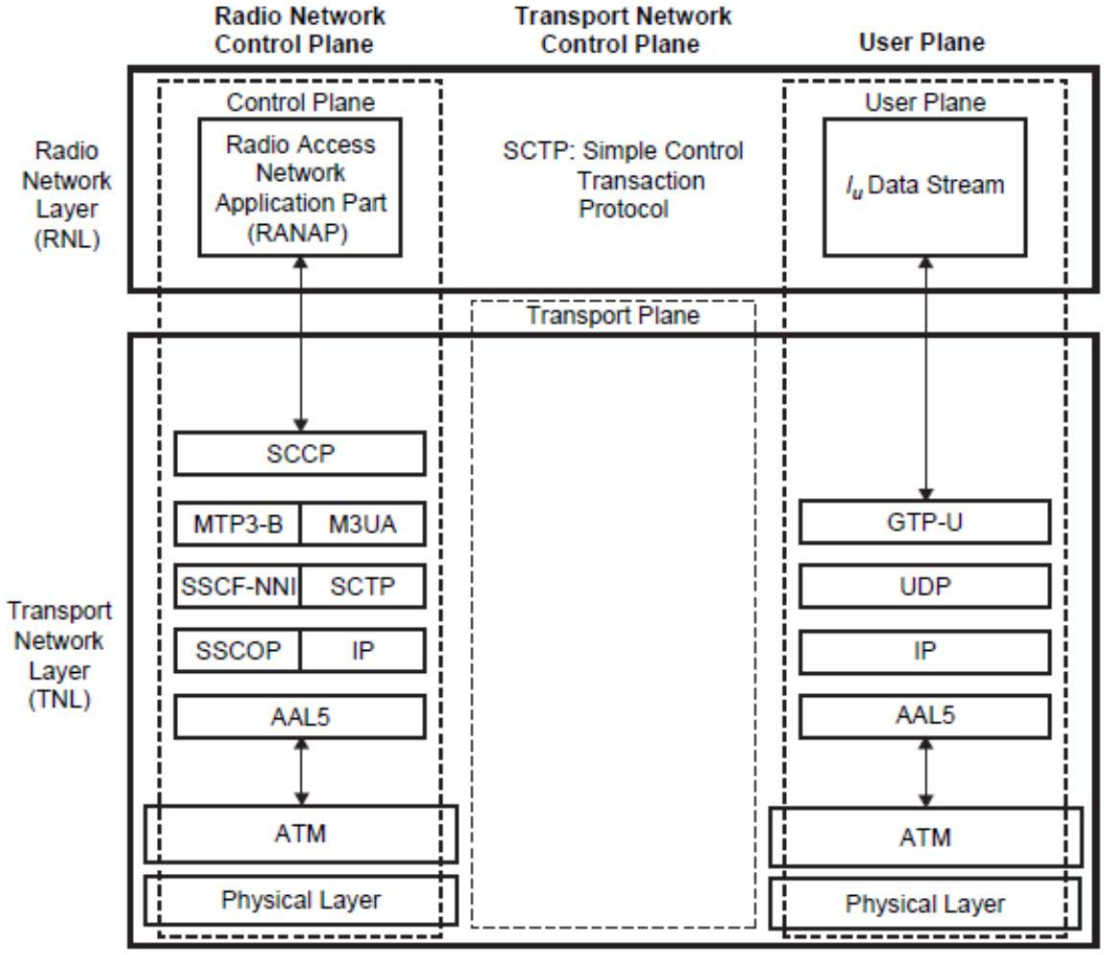


Figure 15.24 PS protocol architecture on I_u interface.

and *ATM adaptation layer 5* (*AAL5*) layers. The SSCF and SSCOP layers are specifi cally designed for signaling transport in ATM networks, and take care of signaling connection management functions. AAL5 is used for segmenting the data to ATM cells.

As an alternative, an IP-based signaling bearer is specifi ed for the *Iu* PS control plane. The IP-based signaling bearer consists of *SS7-MTP3—user adaptation layer* (*M3UA*), *simple control transmission protocol* (*SCTP*), IP, and AAL5. The SCTP layer is specifi cally designed for signaling transport on the Internet.

The *transport network control plane* (*TNCP*) carries information for the

control of transport network used within UCN

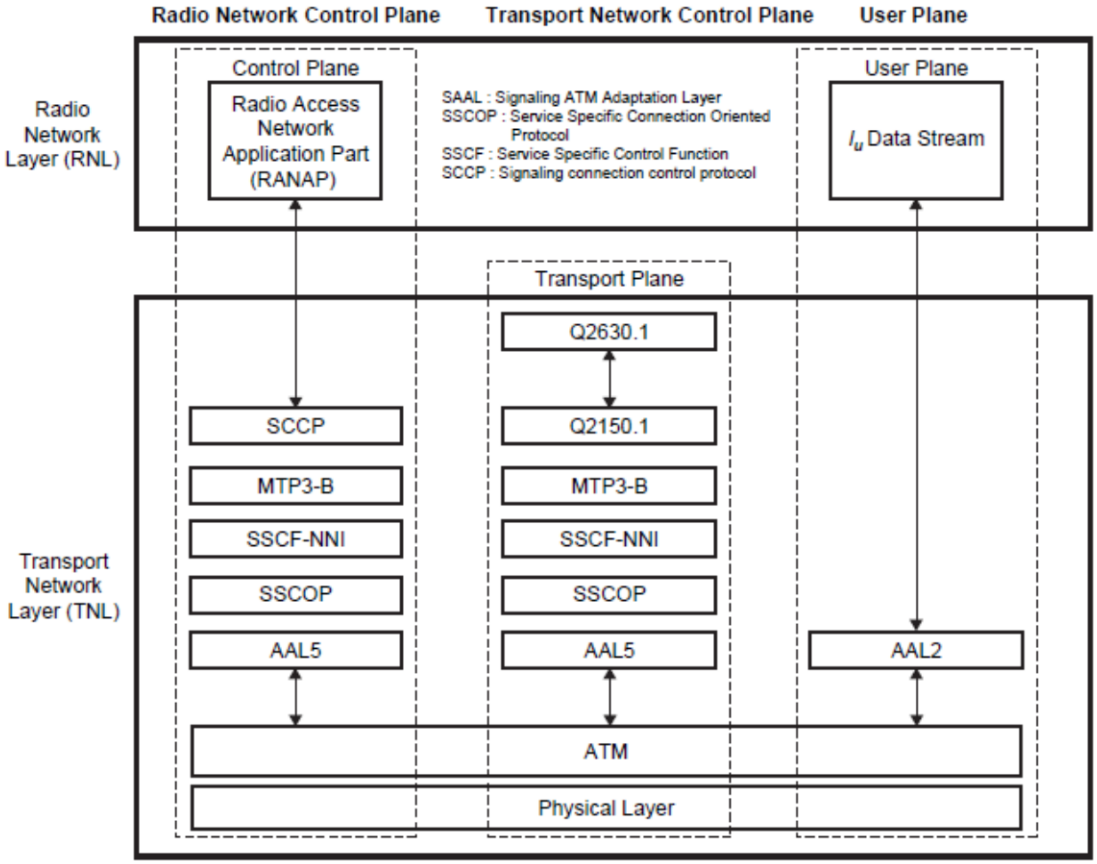


Figure 15.25 CS protocol architecture on I_u interface.

The *user plane* (*UP*) carries user voice and packet data information. AAL2 is used for the following services: narrowband speech (e.g., EFR, AMR); unrestricted digital information service (up to 64 kbps, i.e., ISDN B channel); any low to average bit rate CS service (e.g., modem service to/from PSTN/ISDN). AAL5 is used for the following services: non-real-time PS data service (i.e., best effort packet access) and real-time PS data.

Iur Interface

The connection between two RNCs (serving RNC (SRNC) and drift RNC (DRNC)) is the *Iur* interface. It is used in soft handoff scenarios when different macro diversity streams of one communication are supported by Node Bs that belong to different RNCs. Communication between one RNC and one Node B of two different RNCs are realized through the *Iur* interface. Three different protocol planes are defi ned for it:

Radio network control plane (RNCP)

Transport network control plane (TNCP)

User plane (UP)

The *Iur* interface is used to carry:

Information for the control of radio resources in the context of specifi c service request of one mobile on RNCP, Information for the control of the transport network used within UTRAN

on TNCP, User voice and packet data information on UP

The protocols used on this interface are:

Radio access network application part (RANAP)

DCH frame protocol (DCHFP)

RACH frame protocol (RACHFP)

FACH frame protocol (FACHFP)

Access link control application part (ALCAP)

Q.aal2

Signaling connection control part (SCCP)

Message transfer part 3-B (MTP3-B)

Signaling ATM adaptation layer for network-to-network interface (SAALNNI)

(SAAL-NNI is further divided into service specifi c coordination function for network to network interface (SSCF-NNI), service specifi c connection oriented protocol (SSCOP), and ATM adaptation layer 5 (AAL5))

The bearer is AAL2. The protocol structure of the *Iur* interface is shown in Figure 15.26. Initially, this interface was designed to support the inter-RNC soft handoff, but more features were added during the development of the standard. The *Iur* provides the following four functions:

1. Basic inter-RNC mobility support

Support of SRNC relocation

Support of inter-RNC cell and UTRAN registration area update

Support of inter-RNC packet paging

Reporting of protocol errors

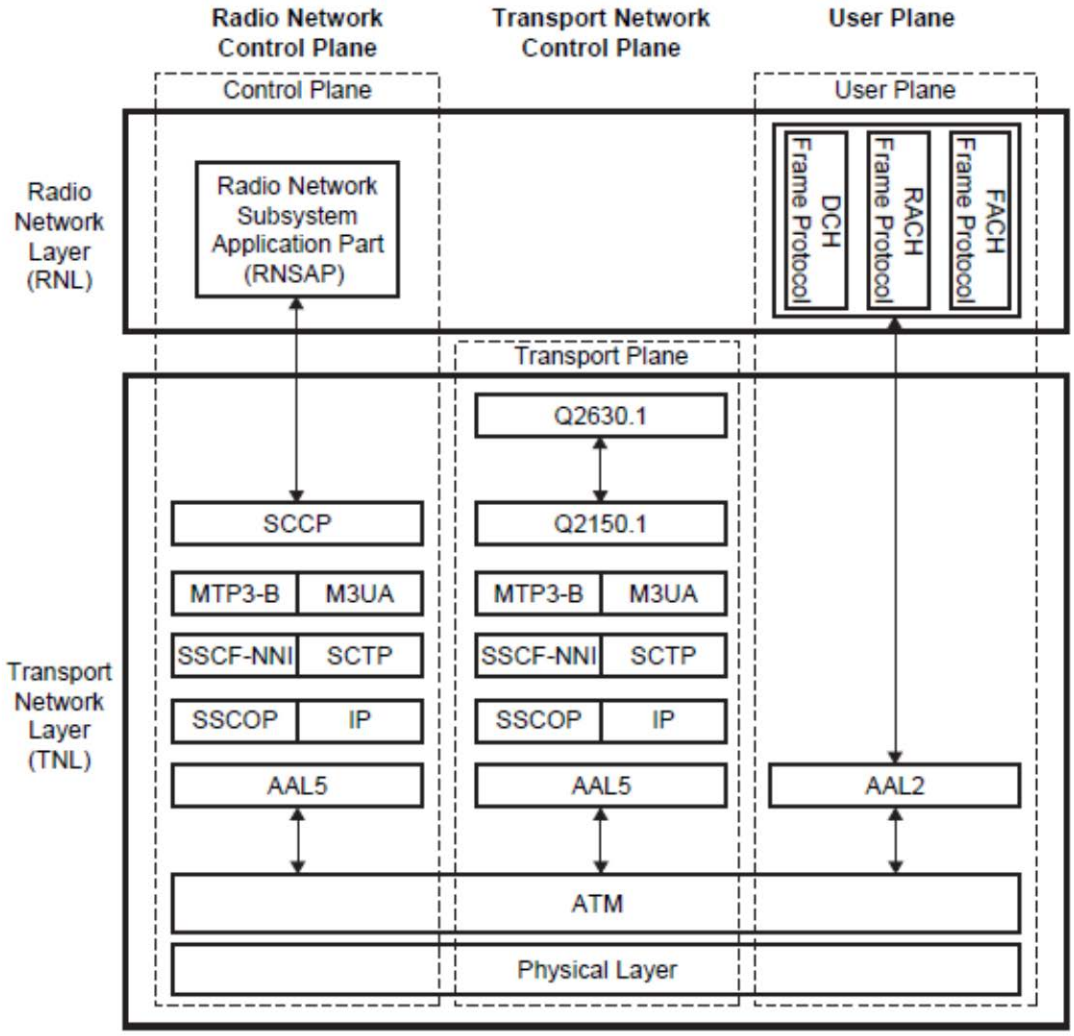


Figure 15.26 Protocol structure of I_{ur} interface.

2.

Dedicated channel traffi c support

Establishment, modifi cation, and release of a dedicated channel in the DRNC due to hard and soft handoff in the dedicated channel state

Setup and release of dedicated transport connections across the *Iur* interface

Transfer of DCH transport blocks between SRNC and DRNC

Management of radio links in the DRNS via dedicated measurement report procedures and power setting procedures

3. Common channel traffi c support

Setup and release of the transport connection across the *Iur* for common channel data streams

Splitting of the MAC layer between the SRNC (MAC-d) and DRNC (MAC-c and MAC-sh); the scheduling for downlink data transmission is performed in the DRNC

Flow control between the MAC-d and MAC-c/MAC-sh

4. Global resource management support

Transfer of cell measurements between two RNCs

Transfer of Node B timing between two RNCs

***Iub* Interface**

The connection between the RNC and Node B is the *Iub* interface. There is one *Iub* interface for each Node B. The *Iub* interface is used for all of the communications between Node B and the RNC of the same RNS. Three different protocol planes are defi ned for it.

Radio network control plane (RNCP)

Transport network control plane (TNCP)

User plane (UP)

The *Iub* interface is used to carry:

Information for the general control of Node B for radio network operation on RNCP, Information for the control of radio resources in the context of specifi c service request of one mobile on RNCP, Information for the control of a transport network used within UTRAN on TCNP

User CC and MM signaling message on RNCP

User voice and packet data information on UP

The protocols used on this interface include:

Node B application part protocol (NBAP)

DCH frame protocol (DCHFP)

RACH frame protocol (RACHFP)

FACH frame protocol (FACHFP)

Access link control application part (ALCAP)

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SSCP or TCP and IP

MTP3-B

SAAL-UNI (SSCF-UNI, SSCOP, and AAL5)

When using multiple low-speed links in the *Iub* interface, Node B supportsnverse multiplexing for ATM (IMA).

The bearer is AAL2. The protocol structure for the interface *Iub* is shown in Figure 15.27.

***Uu* Interface**

The UMTS *Uu* interface is the radio interface between a Node B and one ofits UE. The *Uu* is the interface through which UE accesses the fi xed part of thesystem.

Unit V

PART A

1. **How 4G is describeda? (L-1,CO-5)**

   4G is described as MAGIC — **M**obile multimedia, **A**nytime anywhere, **G**lobal mobility support, **I**ntegrated wireless solution, and **C**ustomized personal service. The 4G systems will not only support the next generation mobile services, but also will support the fi xed wireless networks. The 4G systems are about seamlessly integrating terminals, networks, and applications to satisfy increasing user demands.

2. What are the features of 4G systems? (L-2,CO-5)

High usability: anytime, anywhere, and with any technology

Support for multimedia services at low transmission cost

Personalization

Integrated services

3. What are the applications of 4G system? (L-1,CO-5)

The following are some of the applications of the 4G system:

Virtual presence — 4G will provide user services at all times, even if the user is off-site.

Virtual navigation — 4G will provide users with virtual navigation through which a user can access a database of streets, buildings, etc., of a large city. This requires high speed transmission.

Tele-medicine — 4G will support the remote health monitoring of patients via video conference assistance for a doctor at anytime and anywhere.

Tele-geo-processing applications — 4G will combine geographical information systems (GIS) and global positioning systems (GPS) in which a user will get location querying.

Education — 4G will provide a good opportunity to people anywhere in the world to continue their education on-line in a cost-effective manner.

4. What is meant by Multi carrier modulation? (L-1,CO-5)

Multicarrier modulation (MCM) is a derivative of frequency-division multiplexing.It is not a new technology. Forms of multicarrier systems are currently used in DSL modems and digital audio/video broadcast (DAB/DVB). MCM is a baseband process that uses parallel equal bandwidth subchannels to transmit information and is normally implemented with fast Fourier transform (FFT) techniques. MCM's advantages are better performance in the inter-symbol-interference environment, and avoidance of single-frequency interferers. However, MCM increases the peakto-

average ratio of the signal, and to overcome inter-symbol-interference a cyclic extension or guard band must be added to the data.

5. **Write the formula for Signal to Noise ratio? (L-2,CO-5)**

$$(\text{SNR})_{\text{loss}} = 10\log\frac{L_b + L_c - 1}{L_b}\,(\text{dB})$$

6. **What are the four cases for transmission and reception? (L-3,CO-5)**

Single-Input, Single-Output (SISO)

Single-Input, Multiple-Output (SIMO)

Multiple-Input, Single-Output (MISO)

Multiple-Input, Multiple-Output (MIMO)

7. **What is meant by BLAST system? (L-1,CO-5)**

BLAST is a space division multiplexing (SDM)-based MIMO system. It provides the best trade-off between system performance (spectral effi ciency and capacity) and system implementation complexity. The spectral effi ciency of BLAST ranges from 20 to 40 bps/Hz. It uses a zero-forcing (ZF) nonlinear detection algorithm based on a spatial nulling process combined with symbol cancellation to improve system performance/

8. **What are the steps for ZF algorithm? (L-1,CO-5)**

1. *Ordering*: Determine the optimal detection order.

2. *Nulling*: Choose the nulling vector to null out all the weaker transmit signals and obtain the strongest transmit signal.

3. *Slicing*: Detect the estimated value of the strongest signal by slicing to the nearest value in the signal constellation.

4. *Cancellation*: Cancel the effect of the strongest signal from the received signal vector to reduce the detection complexity for the remaining transmit signal. Go to step 2 — nulling process.

9. **What is the goal for cognitive radio**? **(L-1,CO-5)**

The goal of CR is to relieve radio spectrum overcrowding, which actually translates to a lack of access to full radio spectrum utilization.

10. What is SDR? (L-1,CO-5)

Software-defined radios can be reconfigured “on-the-fl y,” i.e., the universal communication device would reconfigure itself appropriately for the environment. It could be a cordless phone one minute, a cell phone the next, a wireless Internet gadget the next, and a GPS receiver the next. Software-defined radios can be quickly and easily upgraded with enhanced features. In fact, the upgrade could be delivered over-the-air. Software-defined radios can talk and listen to multiple channels at the same time

Part B

1. Define the role of 4G system.(L-1,CO-5)

With the rapid development of wireless communication networks, it is expected

that fourth-generation (4G) mobile systems will be launched within a decade. 4G mobile systems focus on seamless integration of existing wireless technologies including WWAN, WLAN, and Bluetooth (see Figure 23.1). This is in contrast with 3G, which merely focuses on developing new standards and hardware. The recent convergence of the Internet and mobile radio has accelerated the demand for “Internet in the pocket,” as well as for radio technologies that increase data throughput and reduce the cost per bit. Mobile networks are going multimedia, potentially leading to an explosion in throughput from a few bytes for the short message service (SMS) to a few kilobits per second (kbps) for the multimedia messaging service (MMS), to several 100 kbps for video content. In addition to wide area cellular systems, diverse wireless transmission technologies are being deployed, including digital audio broadcast (DAB) and digital video broadcast (DVB) for wide-area broadcasting, local multipoint distribution service (LMDS), and multichannel multipoint distribution service (MMDS) for fi xed wireless

access. IEEE 802.11b, 11a, 11g, 11n, and 11h standards for wireless local area networks (WLANs) are extending from the enterprise world into public and residential domains. Because they complement cellular networks, these new wireless network technologies and their derivatives may well prove to be the infrastructure components of the future 4G mobile networks when multistandard terminals become widely available. This is already the case for WiFi in the public "hotspots," which is being deployed by mobile operators around the world with the aim to offer seamless mobility with wireless wide-area networks.

The 4G systems will encompass all systems from various networks, public to private, operator-driven broadband networks to personal areas, and ad hoc networks. The 4G systems will be interoperable with 2G and 3G systems, as well as with digital (broadband) broadcasting systems. The 4G intends to integrate from satellite broadband to high altitude platform to cellular 2G and 3G systems to wireless local loop (WLL) and broadband wireless access (BWA) to WLAN, and wireless personal area networks (WPANs), all with IP as the integrating

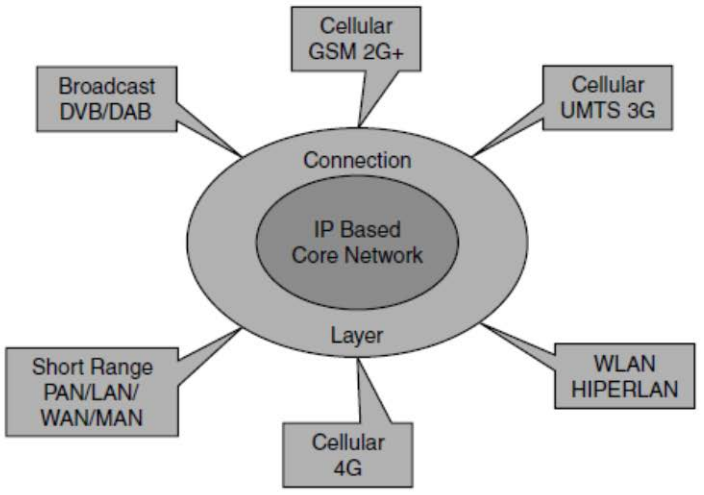


Figure 23.1 Seamless connections of networks.

mechanism. Table provides a comparison of key parameters of 4G with 3G systems.

4G Vision

The 4G systems are projected to solve the still-remaining problems of 3G systems . They are designed to provide a wide variety of new services, from high-quality voice to high-defi nition video to high-data-rate wireless channels. The term 4G is used broadly to include several types of broadband wireless access communication systems, not only cellular systems. 4G is described as MAGIC — **M**obile multimedia, **A**nytime anywhere, **G**lobal mobility support, **I**ntegrated wireless solution, and **C**ustomized personal service. The 4G systems will not only support the next generation mobile services, but also will support the fi xed wireless networks. The 4G systems are about seamlessly integrating terminals, networks, and applications to satisfy increasing user demands.

Accessing information anywhere, anytime, with a seamless connection to a wide range of information and services, and receiving a large volume of information, data, pictures, video, and so on, are the keys of the 4G infrastructure. The future 4G systems will consist of a set of various networks using IP as

Table 23.1 Comparison of key parameters of 4G with 3G.

Details	3G including 2.5G (EDGE)	4G
Major requirement driving architecture	Predominantly voice driven, data was always add on	Converge data and voice over IP
Network architecture	Wide area cell-based	Hybrid-integration of WLAN (WiFi, Bluetooth) and wireless wide-area networks
Speeds	384 kbps to 2 Mbps	20 to 100 Mbps in mobile mode
Frequency band	Dependent on country or continent (1.8 to 2.4 GHz)	Higher frequency bands (2 to 8 GHz)
Bandwidth	5 to 20 MHz	100 MHz or more
Switching design basis	Circuit and packet	All digital with packetized voice
Access technologies	WCDMA, cdma2000	OFDM and multicarrier (MC)-CDMA
Forward error correction	Convolutional codes rate ½, ⅓	Concatenated coding schemes
Component design	Optimized antenna design, multiband adapters	Smart antenna, software-defined multiband and wideband radios
Internet protocol (IP)	Number of airlink protocol including IPv5.0	All IP (IPv6.0)
Mobile top speed	200 km/h	200 km/h

a common protocol. 4G systems will have broader bandwidth, higher data rate,and smoother and quicker handoff and will focus on ensuring seamless service across a multiple of wireless systems and networks. The key is to integrate the 4G capabilities with all the existing mobile technologies through the advanced techniques of digital communications and networking.

4G Features and Challenges

Some key features (primarily from users' points of view) of 4G mobile networks are as follows (see Figure 23.3):

High usability: anytime, anywhere, and with any technology

Support for multimedia services at low transmission cost

Personalization

Integrated services

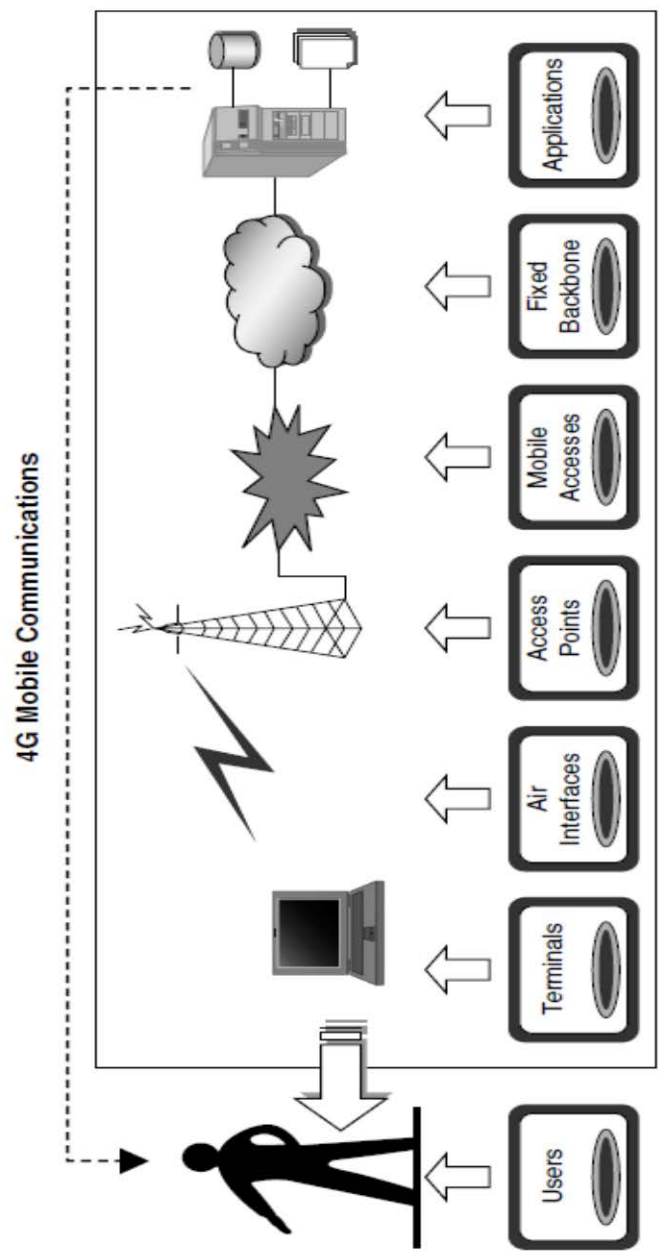


Figure 23.2 4G visions.

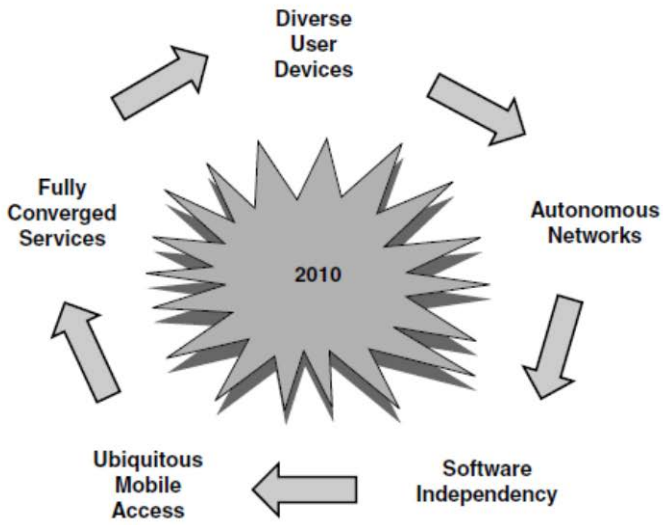


Figure 23.3 4G features.

4G networks will be all-IP-based heterogeneous networks that will allow users to use any system at anytime and anywhere. Users carrying an integrated terminal can use a wide range of applications provided by multiple wireless networks.

4G systems will provide not only telecommunications services, but also data and multimedia services. To support multimedia services, high-data-rate services with system reliability will be provided. At the same time, a low per-bit transmission cost will be maintained by an improved spectral efficiency of the system.

Personalized service will be provided by 4G networks. It is expected that when 4G services are launched, users in widely different locations, occupations, and economic classes will use the services. In order to meet the demands of these diverse users, service providers will design personal and customized service for them. 4G systems will also provide facilities for integrated services. Users can use multiple services from any service provider at the same time. To migrate current systems to 4G with the above-mentioned features, we have to face a number of challenges. Table lists the key challenges and their proposed solutions. Figure shows the carriers migration from 3.5G to 4Gsystems.

Table 23.2 4G Key challenges and their proposed solutions.

	Key challenges	Proposed solutions
Mobile Station		
Multimode user terminals	To design a single user terminal that can operate in different wireless networks, and overcome design problems such as limitations in device size, cost power consumption, and backward compatibilities to systems	A software-defined radio approach can be used: the user terminal adapts itself to the wireless interfaces of the networks.
Wireless system discovery	To discover available wireless systems by processing the signals sent from different wireless systems (with different access protocols and incompatible with each other)	User- or system-initiated discoveries, with automatic download of software modules for different wireless systems
Wireless system selection	Every wireless system has its unique characteristics and role. The proliferation of wireless technologies complicates the selection of the most suitable technology for a particular service at a particular time and place.	The wireless system can be selected according to the best possible fit of user QoS requirements, available network resources, or user preferences.
System		
Terminal mobility	To locate and update the locations of the terminals in various systems. Also, to perform *horizontal* (within the same system) and *vertical* (within different systems) handoff as required with minimum handover latency and packet loss	Signaling schemes and fast handoff mechanisms are proposed.
Network infrastructure and QoS support	To integrate the existing non-IP-based and IP-based systems, and to provide QoS guarantee for end-to-end services that involves different systems	A clear and comprehensive QoS scheme for the UMTS system has been proposed. This scheme also supports interworking with other common QoS technologies.
Security	The heterogeneity of wireless networks complicates the security issue. Dynamic reconfigurable, adaptive, and lightweight security mechanisms should be developed	Modifications in existing security schemes may be applicable to heterogeneous systems. Security handoff support for application sessions is also proposed.

(*Continued*)

	Key challenges	Proposed solutions
Fault tolerance and survivability	To minimize the failures and their potential impacts in any level of tree-like topology in wireless networks.	Fault-tolerant architectures for heterogeneous networks and failure recovery protocols are proposed.
Service		
Multioperators and billing system	To collect, manage, and store the customers' accounting information from multiple service providers. Also, to bill the customers with simple but detailed information.	Various billing and accounting frameworks are being proposed to achieve this goal.
Personal mobility	To provide seamless personal mobility to users without modifying the existing servers in heterogeneous systems.	Personal mobility frameworks are proposed. Most of them use mobile agents, but some do not.

Applications of 4G

The following are some of the applications of the 4G system:

Virtual presence — 4G will provide user services at all times, even if the user is off-site.

Virtual navigation — 4G will provide users with virtual navigation through which a user can access a database of streets, buildings, etc., of a large city. This requires high speed transmission.

Tele-medicine — 4G will support the remote health monitoring of patients via video conference assistance for a doctor at anytime and anywhere.

Tele-geo-processing applications — 4G will combine geographical information systems (GIS) and global positioning systems (GPS) in which a user will get location querying.

Education — 4G will provide a good opportunity to people anywhere in the world to continue their education on-line in a cost-effective manner.

4G Technologies

Multicarrier Modulation

Multicarrier modulation (MCM) is a derivative of frequency-division multiplexing.It is not a new technology. Forms of multicarrier systems are currently used in DSL modems and digital audio/video broadcast (DAB/DVB). MCM is a baseband

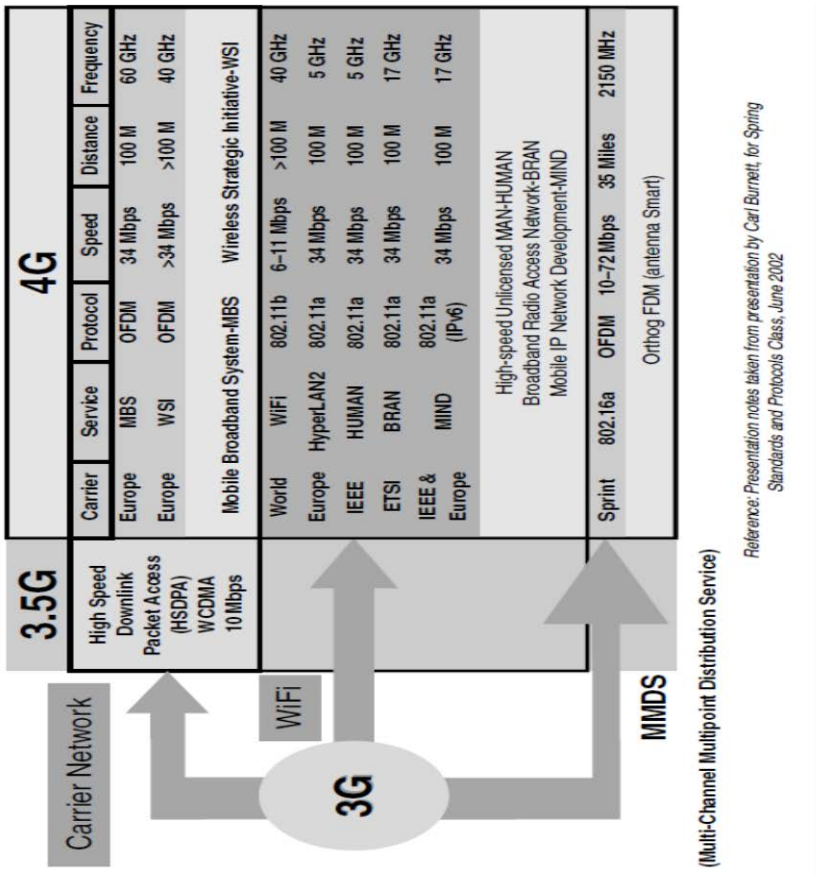


Figure 23.4 Carrier migration from 3.5G to 4G.

process that uses parallel equal bandwidth subchannels to transmit information and is normally implemented with fast Fourier transform (FFT) techniques. MCM's advantages are better performance in the inter-symbol-interference environment, and avoidance of single-frequency interferers. However, MCM increases the peakto- average ratio of the signal, and to overcome inter-symbol-interference a cyclic extension or guard band must be added to the data. The difference, *D*, of the peak-to-average ratio between MCM and a single carrier system is a function of the number of subcarriers, *N*, as:

D(dB) _ 10 log *N* (23.1)

Any increase in the peak-to-average ratio of a signal requires an increase in linearity of the system to reduce distortion. Linearization techniques can be used, but they increase the cost of the system.

If *Lb* is the original length of block, and the channel's response is of length *Lc*, the cyclically extended symbol has a new length *Lb* _ *Lc* _ 1. The new symbol of length *Lb*

_ *Lc* _ 1 sampling periods has no inter-symbol interference. The cost is an increase in energy and uncoded bits are added to the data. At the MCM receiver, only *Lb* samples are processed and *Lc* _ 1 samples are discarded, resulting in a loss in signal-to-noise ratio (SNR) as:

$$(\mathrm{SNR})_{\mathrm{loss}} = 10 \log \frac{L_b + L_c - 1}{L_b} \,(\mathrm{dB})$$

Two different types of MCM are likely candidates for 4G. These include multicarrier code division multiple access (MC-CDMA) and orthogonal frequency division multiplexing (OFDM) using time division multiple access (TDMA). MC-CDMA is actually OFDM with a CDMA overlay. Similar to single-carrier CDMA systems, the users are multiplexed with orthogonal codes to distinguish users in MC-CDMA. However, in MC-CDMA, each user can be allocated several codes, where the data is spread in time or frequency. Either way, multiple users simultaneously access the system. In OFDM with TDMA, the users are assigned time slots to transmit and receive data. Typically MC-CDMA uses quadrature phase shift keying (QPSK) for modulation, while OFDM with TDMA could use more high-level modulations, such as multilevel quadrature amplitude modulation (M-QAM) (where M _ 4 to 256). However, to optimize overall performance, adaptive modulation can be used, where the level of quadrature amplitude modulation (QAM) for all subcarriers is chosen based on measured parameters. In OFDM the subcarrier pulse shape is a square wave. The task of pulse forming and modulation is performed by a simple inverse fast Fourier transform (IFFT) which can be implemented very effi ciently. To decode the transmission, a receiver needs only to implement FFT.

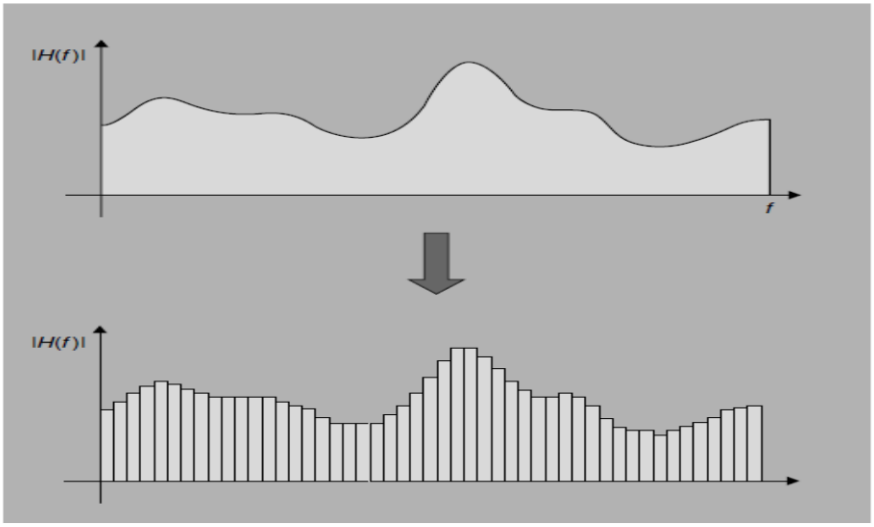


Figure 23.5 A broadband channel divided into many parallel narrowband channels.

The OFDM divides a broadband channel into many parallel subchannels. The subchannel pulse shape is a square wave (see Figure 23.5). The OFDM receiver senses the channel and corrects distortion on each subchannel before the transmitted data can be extracted. In OFDM, each of the frequencies is an integer multiple of a fundamental frequency. This ensures that even though subchannels overlap, they do not interfere with each other (see Figure).

2. What is a multi-input multi-output (MIMO) system? Explain (L-1,CO-5)

Smart Antenna Techniques

Smart antenna techniques, such as multiple-input multiple-output (MIMO) systems, can extend the capabilities of the 3G and 4G systems to provide customers with increased data throughput for mobile high-speed data applications. MIMO systems use multiple antennas at both the transmitter and receiver to increase the capacity of the wireless channel (see Figure . With MIMO systems, it may be possible

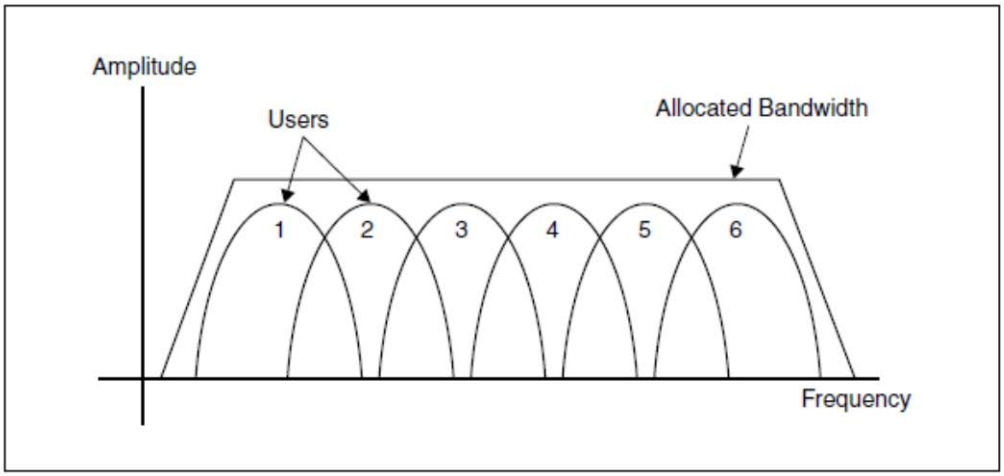


Figure 23.6 Overlapping subchannels.

to provide in excess of 1 Mbps for 2.5G wireless TDMA EDGE and as high as 20 Mbps for 4G systems. With MIMO, different signals are transmitted out of each antenna simultaneously in the same bandwidth and then separated at the receiver. With four antennas at the transmitter and receiver this has the potential to provide four times the data rate of a single antenna system without an increase in transmits power or bandwidth. MIMO techniques can support multiple independent channels in the same bandwidth, provided the multipath environment is rich enough. What this means is that high capacities are theoretically possible, unless there is a direct line of- sight between the transmitter and receiver.

The number of transmitting antennas is *M*, and the number of receiving antennas is *N*, where *N* _ *M*. We examine four cases:

Single-Input, Single-Output (SISO)

Single-Input, Multiple-Output (SIMO)

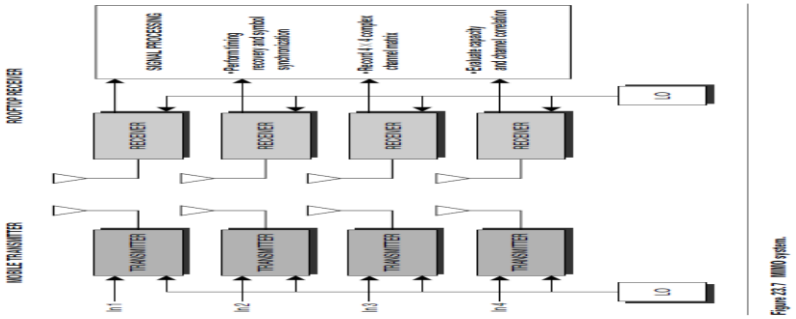
Multiple-Input, Single-Output (MISO)

Multiple-Input, Multiple-Output (MIMO)

Single-input, single-output: The channel bandwidth is *B*, the transmitter power is *Pt*, the signal at the receiver has an average signal-to-noise ratio of SNR0, then the Shannon limit on channel capacity *C* is

C _ *B* log 2 (1 _ SNR 0) (23.3)

Fig :Overlapping subchannels.



Single-input, multiple-output: There are *N* antennas at the receiver. If the signals received on the antennas have on average the same amplitude, then they can be added coherently to produce an *N*2 increase in signal power. There are *N* sets of noise sources that are added coherently and result in an *N*-fold increase in noise power. Hence, the overall increase in SNR will be:

$$\text{SNR} \approx \frac{N^2 \times (\text{signal power})}{N \times (\text{noise})} = N \times \text{SNR}_0$$

The capacity for this channel is approximately equal to

$$C \approx B \log_2 [1 + N \times \text{SNR}_0]$$

Multiple-input, single-output: We have *M* transmitting antennas. The total power is divided into *M* transmitter branches. If the signals add coherently at the receiving antenna, we get an *M*-fold increase in SNR as compared to SISO. Because there is only one receiving antenna, the noise level is same as SISO. The overall increase in SNR is approximately

$$\text{SNR} \approx \frac{M^2 \cdot [(\text{signal power})/M]}{\text{noise}} = M \times \text{SNR}_0$$

Multiple-input, multiple-output: MIMO systems can be viewed as a combination of MISO and SIMO channels. In this case, it is possible to achieve approximately an *MN*-fold increase in the average SNR0 giving a channel capacity equal to

$$C \approx B \log_2(1 + M \times N \times \text{SNR}_0)$$

Assuming *N* _ *M*, we can send different signals using the same bandwidth and still be able to decode correctly at the receiver. Thus, we are creating a channel for each one of the transmitters. The capacity of each one of these channels is roughly equal to

$$C_{\text{single}} \approx B \log_2\left(1 + \frac{N}{M} \times \text{SNR}_0\right)$$

Since we have *M* of these channels (*M* transmitting antennas), the total capacity of the system is

$$C \approx MB \log_2\left(1 + \frac{N}{M} \times \text{SNR}_0\right)$$

We get a linear increase in capacity with respect to the transmitting antennas. As an example we assume SNR0 is equal to 10 dB, *M* _ 4, *N* _ 5 and bandwidth *B* (MHz) and list the system capacity for each channel type in Table 23.3.

3. How are higher spectral effi ciency and increased throughput achieved in the OFDM-MIMO system? (L-3,CO-5)

OFDM and MIMO techniques can be combined to achieve high spectral efficiency and increased throughput. The OFDM-MIMO system transmits independent OFDM modulated data from multiple antennas simultaneously. At the receiver, after OFDM

demodulation, MIMO decodes each subchannel to extract data from all transmit antennas on all the subchannels.

Adaptive Modulation and Coding with Time-Slot Scheduler

In general, TCP/IP is designed for a highly reliable transmission medium in wired networks where packet losses are seldom and are interpreted as congestion in the network [1]. On the other hand, a wireless network uses a time varying channel where packet losses may be common due to severe fading. This is misinterpreted by TCP as congestion which leads to ineffi cient utilization of the available radio link capacity. This results in signifi cant degradation of the wireless system performance. There is a need for a system with effi cient packet data transmission using TCP in 4G [2,3,5]. This can be achieved by using a suitable automatic repeat request (ARQ) scheme combined with an adaptive modulation and coding system, and a time-slot scheduler that uses channel predictions. This way, the lower layers are adapted to channel conditions while still providing some robustness through retransmission. The time-slot scheduler shares the spectrum efficiently between users while satisfying the QoS requirements.

If the channel quality for each radio link can be predicted for a short duration (say about 10 ms) into the future and accessible by the link layer, then ARQ along with an adaptive modulation and coding system can be selected for each user to satisfy the bit error rate (BER) requirement and provide high throughput. The scheduler uses this information about individual data streams (along with predicted values of different radio links and selected modulation and coding systems by the link layer) and distributes the time slots among the users.

Table 23.3 Comparison of channel capacity for different channel types.

Channel type	Capacity (Mbps)	Normalized capacity with respect to SISO
SISO	3.45 B	1.0
SIMO	5.66 B	1.64
MISO	5.35 B	1.55
MIMO (with same input)	7.64 B	2.21
MIMO (with different input)	15 B	4.35

The planning is done so that the desired QoS and associated priority to different users are guaranteed while channel spectrum is efficiently utilized.

4.What is Bell Labs Layered Space Time (BLAST) System & Explain (L-1,CO-5)

BLAST is a space division multiplexing (SDM)-based MIMO system. It provides the best trade-off between system performance (spectral effi ciency and capacity) and system implementation complexity. The spectral effi ciency of BLAST ranges from 20 to 40 bps/Hz. It uses a zero-forcing (ZF) nonlinear detection algorithm based on a spatial nulling process combined with symbol cancellation to improve system performance. The BLAST exploits multipath by using scattering characteristics of the propagation environment to enhance transmission accuracy. Figure 23.8 shows the architecture of the BLAST system.

Transmitter: The data stream of a user is divided into multiple substreams. An array of transmit antennas (*M*) is used to simultaneously launch parallel data substreams. Each substream is mapped to a symbol by the same constellation and sent to its transmit antenna. All substreams are transmitted in the same frequency band and are independent of one another. Effective transmission rate is increased roughly in proportion to the number of transmit antennas used. The individual transmitter power is scaled by 1/*M*, so that the total power remains constant independent of the number of transmitters.

Receiver: An array of antennas (*N* _ *M*) is used to receive multiple transmitted substreams and their scattered images. Since substreams originate from different transmit antennas, they are located at different points in space. Using sophisticated signal processing, the substreams are identifi ed and recovered.

Model: Each time sequence *sj*(*t*), *j* _ 1, 2, . . . , *M* is referred to as a layer. At the receiver, the signal *ri*(*t*) is received at time *t*. It is a noisy superposition of the *M* transmitted signal respectively corrupted by noise *ni*(*t*):

$$r_i(t) = \sum_{j=1}^{M} h_{ij}(t)s_i(t) + n_i(t) \quad i = 1, 2, 3, \ldots, N$$

where:

hij(*t*) is the channel gain (complex transfer function) from transmit antenna *j* to receive antenna *i* at any time *t*

We make the following assumptions:

Quasi-static flat fading channel. That is, channel gain *hij*(*t*) remains constant over a block of time, and then changes block by block in an independent manner. Channel is rich scattering. This is true if antenna spacing is suffi cient (i.e., several times of wavelength). This condition provides a large number of

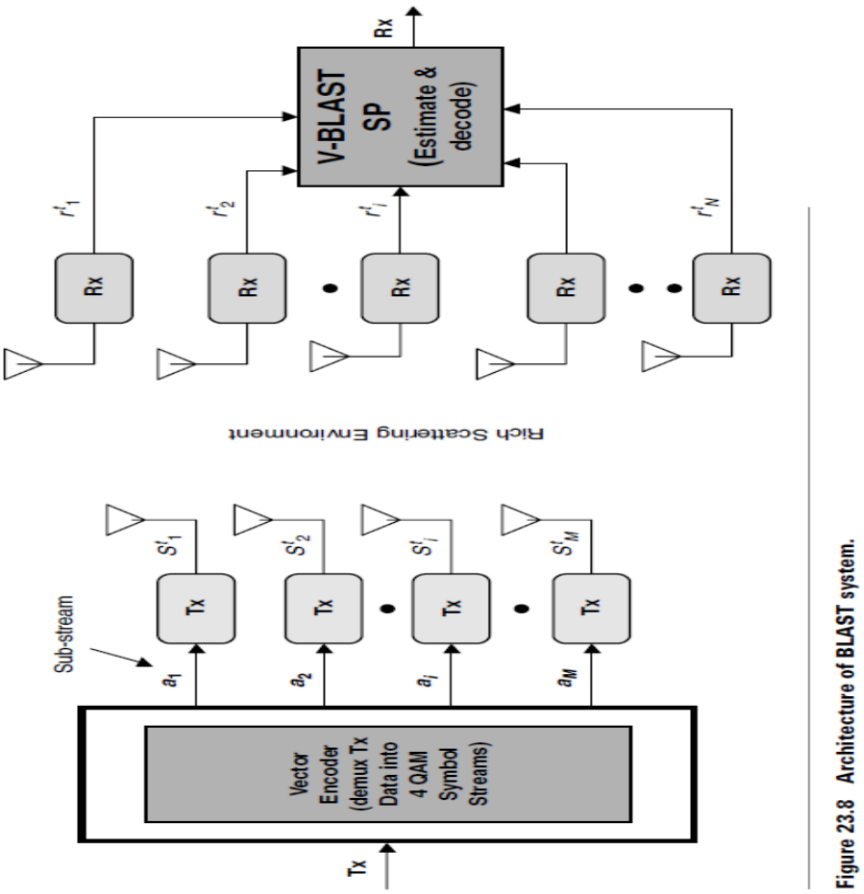


Figure 23.8 Architecture of BLAST system.

local scatters around transmitter or receiver and supports that the channel gains are complex Gaussian and independent of one another.Equation 23.10 can be written as:

$$[r] = [H] \cdot [s] + [n]$$

$$[r] = \begin{bmatrix} r_1 \\ r_2 \\ \bullet \\ \bullet \\ r_N \end{bmatrix}; [n] = \begin{bmatrix} n_1 \\ n_2 \\ \bullet \\ \bullet \\ n_N \end{bmatrix}; [s] = \begin{bmatrix} s_1 \\ s_2 \\ \bullet \\ \bullet \\ s_M \end{bmatrix} \text{ and } [H] = \begin{bmatrix} h_{11} & h_{12} & \bullet & \bullet & h_{1M} \\ h_{21} & h_{22} & \bullet & \bullet & h_{2M} \\ \bullet & \bullet & \bullet & \bullet & \bullet \\ \bullet & \bullet & \bullet & \bullet & \bullet \\ h_{N1} & h_{N2} & \bullet & \bullet & h_{NM} \end{bmatrix}$$

Signal Processing Algorithm: At the bank of the receiving antennas, highspeed signal processors look at signals from all the receiving antennas simultaneously, first extracting the strongest substream from the morass, then proceeding with the remaining weaker signals, which are easier to recover once the stronger signals have been removed as a source of interference. Maximum-Likelihood (ML) detection is optimal for BLAST, but it is too complex to implement. As an example, with six transmit antennas and QPSK modulation, a total of 46 _ 4096 comparisons have to be made for each transmitted symbol. A low complexity suboptimal detection algorithm, called ZF is used. At each symbol time, the strongest layer (transmitted signal) is fi rst detected and its effect is cancelled for each

received signal. We then proceed to detect the strongest of the remaining layers, and so on. The ZF algorithm consists of four recursive steps:

1. *Ordering*: Determine the optimal detection order.
2. *Nulling*: Choose the nulling vector to null out all the weaker transmit signals and obtain the strongest transmit signal.
3. *Slicing*: Detect the estimated value of the strongest signal by slicing to the nearest value in the signal constellation.
4. *Cancellation*: Cancel the effect of the strongest signal from the received signal vector to reduce the detection complexity for the remaining transmit signal. Go to step 2 — nulling process.

Implementation: Each transmitter is a QAM transmitter and operates with synchronized symbol timing, i.e., the collection of transmitters comprises a vector

valued transmitter. 16-QAM signal constellation is used for each transmitter. The total transmitted power is kept constant. Blast algorithm in 3 different architectures 4 _ 8, 8 _ 12, and 12 _ 16 have been used. Results are shown in Figures 23.9, 23.10, and 23.11.

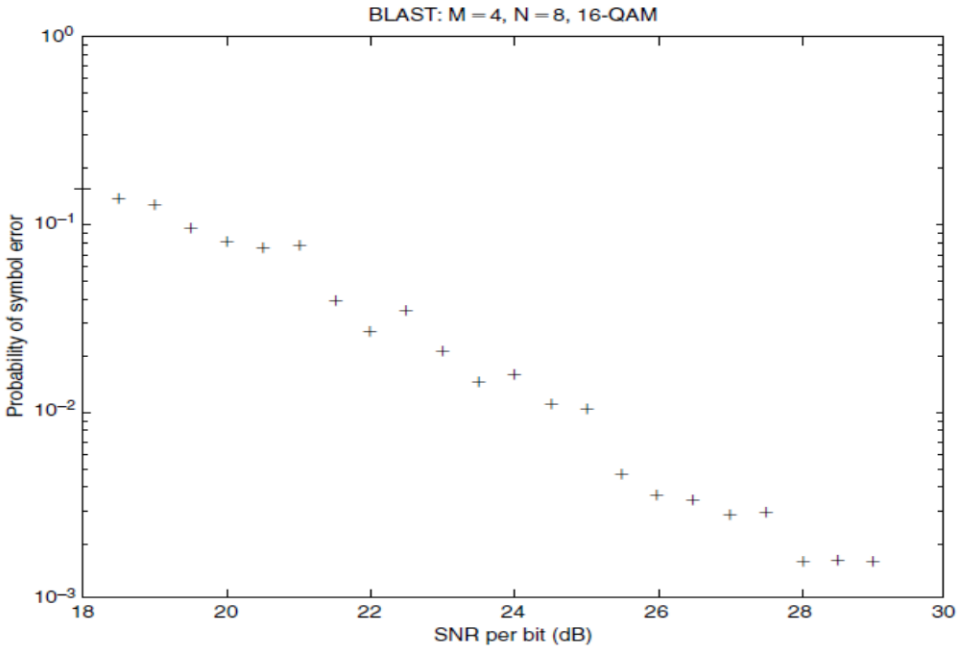


Figure 23.9 BLAST 4 × 8.

BLAST shows promising results in enhancing spectral efficiency. Further gain in spectral efficiency can be obtained by using high-level M-QAM and OFDM.

5.Explain about Software-Defined Radio. (L-1,CO-5)

A software-defined radio (SDR) system is a radio communication system which uses software for the modulation and demodulation of radio signals. An SDR performs significant amounts of signal processing in a general purpose computer, or a reconfigurable piece of digital electronics. The goal of this design is to produce a radio that can receive and transmit a new form of radio protocol just by running new software. Software-defined radios have significant utility for cell phone services, which must serve a wide variety of changing radio protocols in real time. The hardware of a software-defined radio typically consists of a super heterodyne RF front end which converts RF

signals from and to analog RF signals, and analog to digital converters and digital to analog converters which are used to convert digitized intermediate

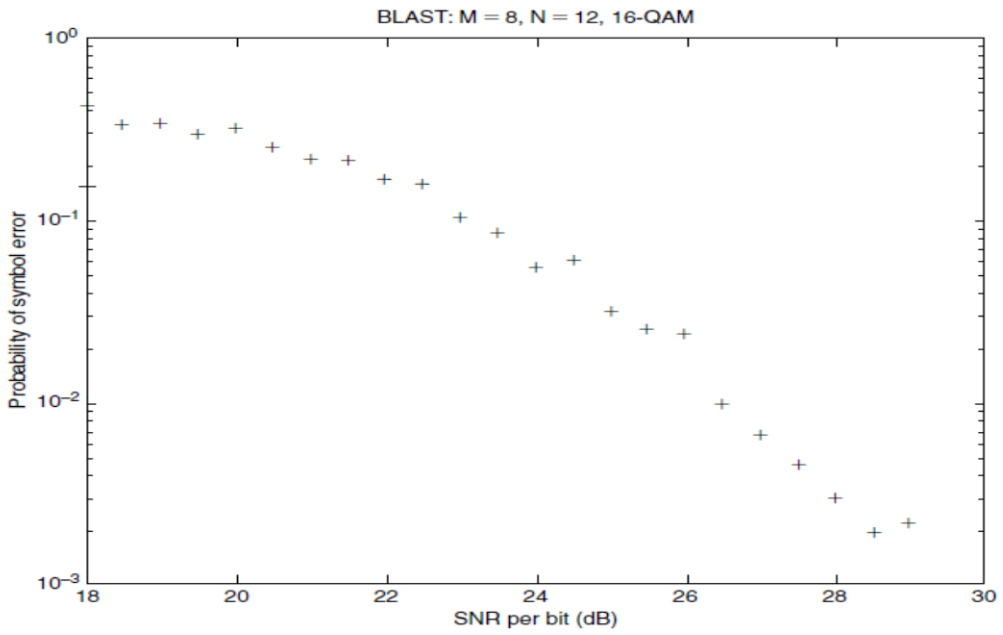


Figure 23.10 BLAST 8 × 12.

frequency (IF) signals from and to analog form, respectively. Software-defined radio can currently be used to implement simple radio modem technologies. In the long run, SDR is expected to become the dominant technology in radio communications. The following are some of the things that SDR can do that haven't been possible before:

Software-defined radios can be reconfigured "on-the-fl y," i.e., the universal communication device would reconfigure itself appropriately for the environment. It could be a cordless phone one minute, a cell phone the next, a wireless Internet gadget the next, and a GPS receiver the next. Software-defined radios can be quickly and easily upgraded with enhanced features. In fact, the upgrade could be delivered over-the-air. Software-defined radios can talk and listen to multiple channels at the same time.

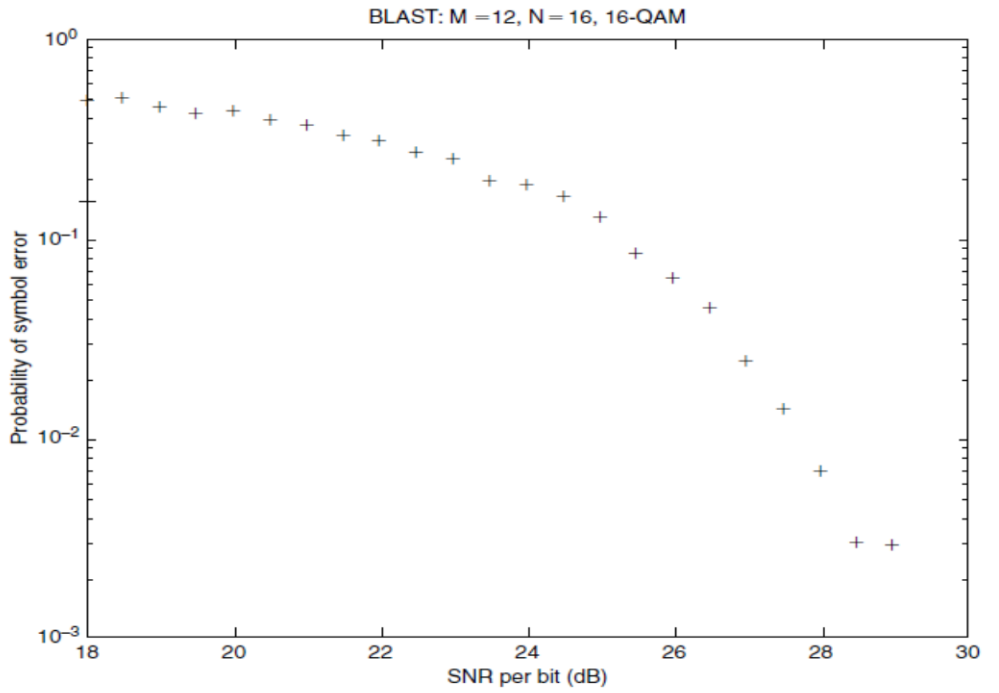


Figure 23.11 BLAST 12 × 16.

New kinds of radios can be built that have never before existed. Smart radios or cognitive radios (CRs) can look at the utilization of the RF spectrum in their immediate neighbourhood and configure themselves for the best performance.

6.What is Cognitive Radio? (L-1,CO-5)

With the CR paradigm, spectrum can be efficiently shared in a more flexible fashion by a number of operators/users/systems. The CR can be viewed as an enabling technology that will benefi t several types of users by introducing new communications and networking models for the whole wireless world, creating better business opportunities for the incumbent operators and new technical dimensions for smaller operators, and helping shape an overall more efficient approach regarding spectrum requirements and usage in the next generation wireless networks.

The CR can be regarded as an extension of SDR. In 2003, the IEEE Committee on Communications and Information Policy (CCIP) recommended CR. for consideration by the FCC as a means to conserve valuable spectrum utilization. The CR focuses on applying software capabilities that have been developed to support algorithm control

across a wide spectrum of signal processing technologies to add smarts to the software that allows it to determine when frequencies are free to use and then use them in the most efficient manner possible.

Most of the research work currently is focusing on spectrum sensing cognitive radio — particularly on the utilization of TV bands for communication. The essential problem of spectrum sensing CR is the design of high quality sensing devices and algorithms for exchanging spectrum sensing data between nodes. It has been shown in [6] that a simple energy detector cannot guarantee accurate detection of signal presence. This calls for more sophisticated spectrum sensing techniques and requires that information about spectrum sensing be exchanged between nodes regularly.

It is not implicit that a CR must be software-defi ned radio. It is possible to implement CR features — the ability to detect and avoid (protect) incumbent users — while using relatively conventional radio transmitter/receiver architectures and techniques. The goal of CR is to relieve radio spectrum overcrowding, which actually translates to a lack of access to full radio spectrum utilization.