**S-72.3250**

**Laboratory works in Radiocommunications**

# Investigation of WLAN throughput

# 1 Table of Contents

## 2 Abbreviations

| | |
|---|---|
| ACK | Acknowledgement |
| AP | Access Point |
| CSMA/CA | Collision Sense Multiple Access with Collision Avoidance |
| CTS | Clear to Send |
| DIFS | DCF Interframe Space |
| DSSS | Direct Sequence Spread Spectrum |
| IETF | Internet Engineering Task Force |
| IP | Internet Protocol |
| LAN | Local Area Network |
| LED | Light Emitting Diode |
| LLC | Logical Link Control |
| MAC | Medium Access Control |
| MOS | Mean Opinion Score |
| PCMCIA | Personal Computer Memory Card International Association |
| PDU | Protocol Data Unit |
| PPDU | PLCP PDU |
| PSQM | Perceptual Speech Quality Measure |
| RTS | Request to Send |
| SDU | Service Data Unit |
| SIFS | Short Interframe Space |
| SNR | Signal to Noise Ratio |
| STA | Station |
| UDP | User Datagram Protocol |
| VoIP | Voice over IP |
| WLAN | Wireless Local Area Network |

# 3 Introduction

This laboratory work will introduce the students the IEEE 802.11b Wireless LAN (Local Area Network). The goal of this laboratory work is to understand the principles of the IEE 802.11b WLAN and the parameters that affect its efficiency. This work tries to outline the students the principles of WLAN network planning demonstrating their interference to each other. After the work, the student should know what IEEE 802.11b network is capable of and be able to analyze the effect of changes in e.g. packet size, use of RTS/CTS and interference to an IEEE 802.11b network.

# 4 IEEE 802.11

IEEE 802.11 was approved as an IEEE standard in 1997. The latest version of the standard was published in 1999. It defines a technology that provides wireless connections for stations in a small geographical area. The wireless connections use the Industrial Scientific and Medical (ISM) band at 2.4 GHz.

## 4.1 Architecture

The basic building blocks of the IEEE 802.11 network are Access Points (APs) and Stations (STAs). The AP offers access to the wired network through the wireless medium. The building blocks of IEEE 802.11 can be grouped together as Basic Service Sets (BSSs), which can be either independent or infrastructure type.

In the Independent BSS (IBSS) STAs communicate directly with each other without an AP. In the IBSS the STAs that wish to communicate with each other have to be in each other’s radio range.

The infrastructure BSS consists of an AP and zero or more STAs. In an infrastructure BSS all the traffic has to go through the AP.

The two possible BSSs are described in Figure 1. Several BSS can be connected together using a Distribution System (DS), which can be e.g. a wired connection between the APs. The IEEE 802.11 infrastructure BSSs can yet form a greater entity called an Extended Service Set (ESS). The ESS consists of several BSSs and a DS. The mobility of the STAs under an ESS is hidden to devices outside the ESS.

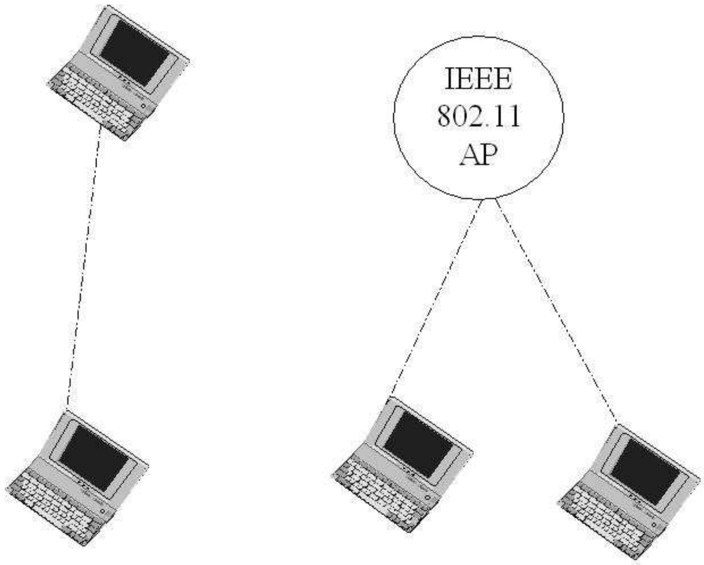


**Figure 1 Independent and infrastructure BSSs.**

The IEEE 802.11 protocol stack consists of two layers: Physical (PHY) and Medium Access Control (MAC). The MAC layer is a sublayer of the data link layer of the OSI model [ISO94]. In the IEEE 802.11 the PHY layer is divided into Physical Layer Convergence Procedure (PLCP) and Physical Medium Dependent (PMD) sublayers. The IEEE 802.11 network can be connected to upper protocol layers using Logical Link Control protocol (LLC) (IEEE 802.2). The LLC belongs to the data link layer of the OSI model, just above the MAC layer.

## 4.2 MAC layer

The MAC layer is a sublayer of the datalink layer of the OSI model. The task of the MAC layer is to insert the data coming from higher layers into frames to be forwarded to the PHY layer. The MAC layer provides an interface to the higher protocol layers. The MAC uses Collision Sense Multiple Access with Collision Avoidance (CSMA/CA) to control the access to the wireless medium. The CSMA scheme is familiar from Ethernet, but whereas the Ethernet uses CD (Collision Detection), IEEE 802.11 uses CA.

The MAC layer uses two kinds of control functions to handle the access to the medium – Distributed Control Function (DCF) and Point Control Function (PCF). The PCF is not widely used [Gas02, pp. 140], but it is specified in the standard [IE$^3$99a, pp. 86]. In DCF the access control to the medium is handled by every STA individually. The idea of PCF is that a point coordinator inside an AP will decide which STA has access at a time.

### *Framing in IEEE 802.11 MAC*

All the higher layer traffic that is transmitted using IEEE 802.11, uses data frames (Figure 2). The other frame types are related to MAC operation and network management tasks [IE$^3$99a, Section 7.2].

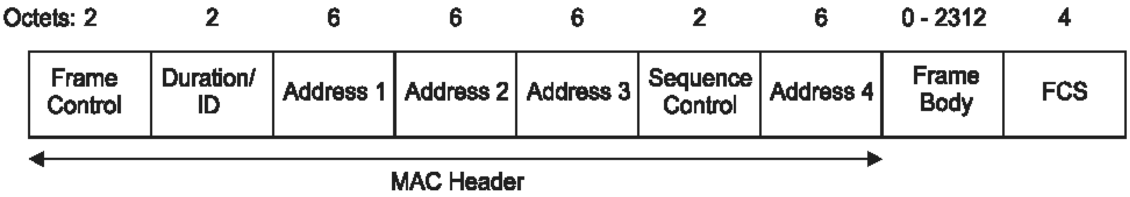


**Figure 2 Data frame [IE$^3$99a, pp. 44]**

The description for each field of a data frame can be found in [IE$^3$99a, Chapter 7]. The information from higher layers (such as IP packets) is carried in the frame body. If the total length of a MAC frame is greater than the fragmentation threshold (see Section: Fragmentation of frames), the frame is fragmented. From Figure 2 can also be seen that the MAC frame contains a Frame Check Sequence (FCS) field for detecting possible errors in the MAC header and frame body. The polynomial used for creating FCS is the same as the one used in all IEEE 802 LAN standards.

Acknowledgement (ACK) frame (Figure 3) is a frequently used control frame in IEEE 802.11. It is sent when a previous frame is received correctly as depicted in Figure 5. The total length of an ACK frame is 14 bytes.

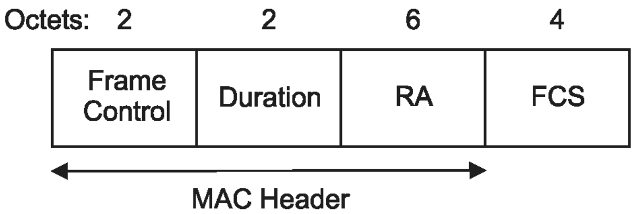


**Figure 3 Acknowledgement frame in IEEE 802.11 [IE$^3$99a, pp. 42].**

### *Fragmentation of frames*

The IEEE 802.11 uses the ISM band, which is in free use. The ISM band is used by e.g. other wireless LAN systems, Bluetooth and microwave ovens. The devices operating in the same frequency band can cause interference to each other.

The traffic in IEEE 802.11 is based on frames. The longer the frame is in bytes the longer it is in time. The retransmission of long frames, due to transmission errors like collisions or microwave-oven interference, is very time consuming and wastes capacity. It can, therefore, be more economical to send short frames, so that the colliding frames wouldn't waste so much capacity. To avoid collisions and the effects of a bad radio channel, a fragmentation threshold parameter can be set in 802.11 MAC. It defines the maximum length of a MAC frame that can be sent without splitting the frame into multiple fragments. If the MAC frame is fragmented, all the fragments are transmitted to the receiver with only two Short Interframe Spaces (SIFSs), and an acknowledgement frame between them, as depicted in Figure 4.

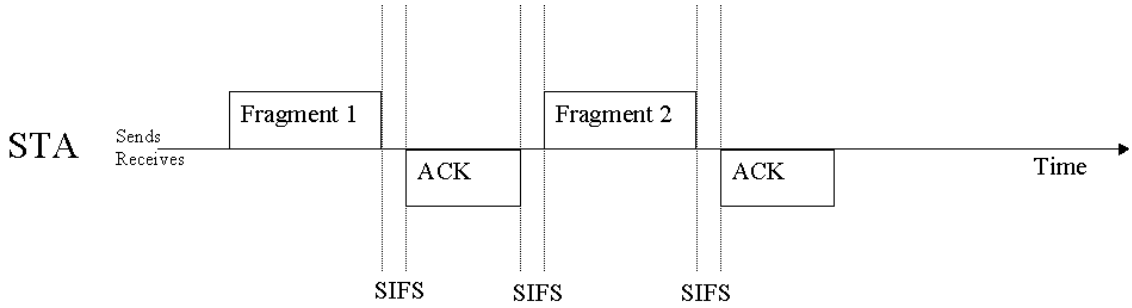


**Figure 4 Fragmentation of a long frame.**

### *DCF*

The DCF aims at that only one STA would use the wireless medium at a time. If several transmissions take place, which are detectable to the receiving STA, the receiver will notice a collision and all the data will be lost without a retransmission. The elementary data transmission in IEEE 802.11 consist of a frame that is sent to the recipient and an acknowledgement that is sent back (Figure 5). All the frames (excluding multicast frames) sent are acknowledged in IEEE 802.11.

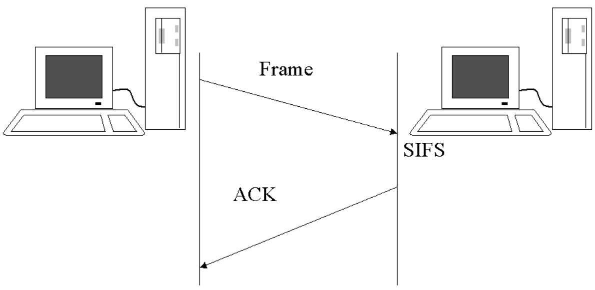


**Figure 5 Elementary data transfer in IEEE 802.11**

The wireless medium is not occupied with traffic all the time. The frame sizes are limited and the frames are separated with spaces between them. These spaces between frames are called Interframe Spaces (IFSs) and there are four of them: Short Interframe Space (SIFS), PCF Interframe Space (PIFS), DCF Interframe Space (DIFS) and Extended Interframe Space (EIFS). The shorter the IFS, the higher the priority for the use of the medium.

#### SIFS

This IFS is the shortest IFS and it is used between acknowledgement frames, CTS frames, subsequent fragments of transmission and responses to polling during Point Coordination Function (PCF) operation.

#### PIFS

This interframe space is used to start the Contention Free (CF) period for PCF operation and during the CF period as the basic IFS in carrier sensing.

DIFS

This IFS is used during DCF for carrier sensing operation. After DIFS period the STAs may use the medium if it is sensed free.

EIFS

This IFS is used by the STA that received a MAC frame incorrectly.

To gain access to the medium the STAs have to contend for it. The contention will take place after the channel has been idle for one DIFS period. Every STA has an equal possibility to gain access to the channel – the traffic is thus best effort traffic. If an STA uses shorter interframe space than one DIFS, its traffic will have a higher priority than the traffic of other STAs. The actual contention is performed using backoff timers. After sensing the medium to be idle for at least one DIFS period of time the medium may be occupied by any STA [IE$^3$99a, pp. 75].

A backoff timer is a counter that is STA specific. The STA will decrement the backoff timer until it reaches zero after which the STA will send its frame. The STA will decrement its backoff timer whenever the channel has been idle for one DIFS period. The backoff timer is generated using a random number generated from a uniform distribution and a Contention Window (CW) that is STA specific. The CW minimum and maximum lengths and the timeslot duration are PHY layer specific and are defined in [IE$^3$99a].

**Channel reservation**

When IEEE 802.11 frames are formed the IEEE 802.11 MAC Protocol Data Unit (MPDU) will have a duration field that indicates how long time the medium will be reserved for traffic. The duration field forms a so-called Network Allocation Vector (NAV) and its use is described in Figure 6.

In high traffic situations it is good to use RTS/CTS for channel reservation (Figure 6). An STA using RTS/CTS will send an RTS frame to the receiving STA, which will respond with a CTS frame. After this RTS/CTS exchange the actual frame containing the payload is sent.

Because RTS and CTS frames are short (20 and 14 bytes respectively), the time wasted in possible collisions is short too. RTS/CTS exchange causes some overhead to the actual transmission. RTS/CTS is studied in [Bin99] in which it is shown that it is effective when compared to plain CSMA/CA when frame sizes are increased. By using RTS/CTS the throughput can be kept almost constant even if the number of STAs gets higher. In [Bin99] it is also shown that the use of RTS/CTS with small frame sizes is not reasonable. The overhead caused by RTS/CTS operation is therefore justified, if the traffic is high and the transmitted frames are long.

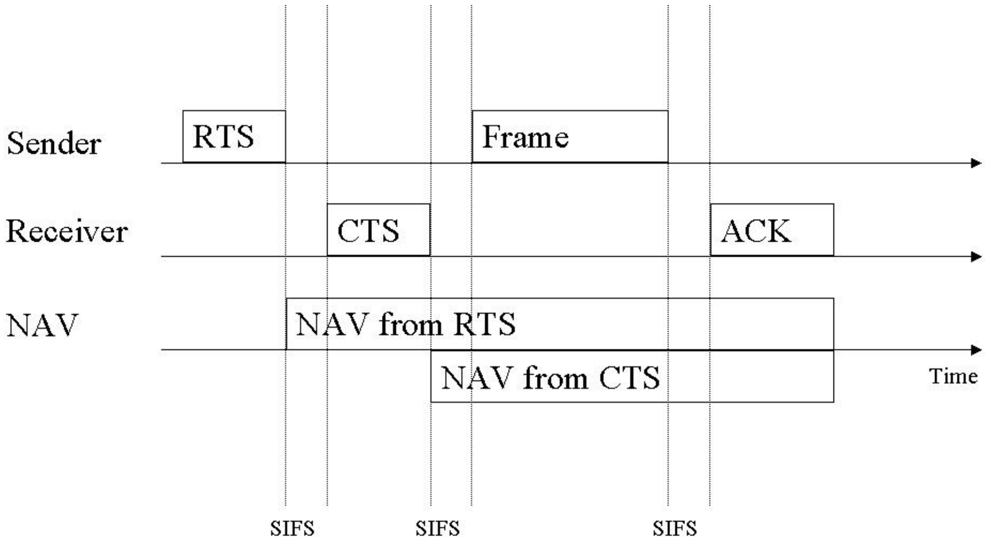


**Figure 6 RTS/CTS exchange with the NAV update.**

### *PCF*

PCF enables contention free services for STAs. PCF is not widely implemented in IEEE 802.11 devices partly because it is an optional feature of IEEE 802.11 standard [Gas02, pp. 140] [IE$^3$99a, pp. 90]. The AP in an infrastructure BSS will have a time window called contention-free period repetition interval which includes operation time for both PCF and DCF (Figure 7).

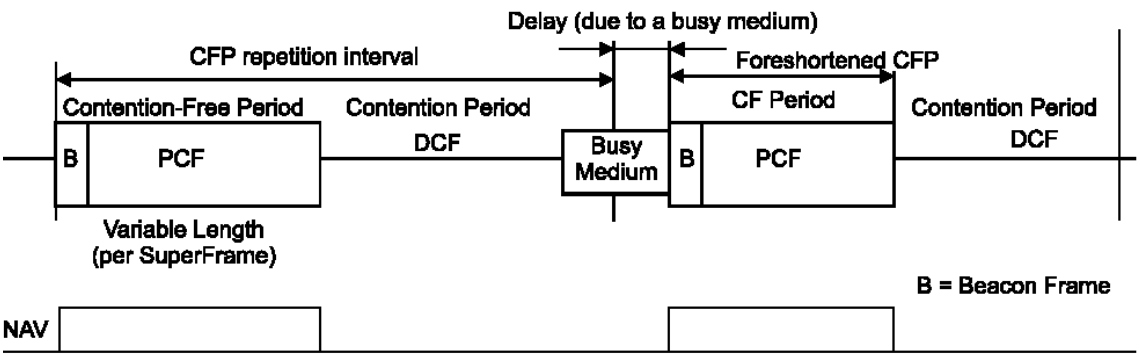


**Figure 7 DCF and PCF operating at the same time.**

The contention-free period repetition interval is further divided into Contention-Free Period (CFP) and Contention Period (CP). The PCF controls the access to the medium during CFP and the DCF during CP. In PCF the AP will have a polling list which indicates which STAs are pollable. During CFP the AP will poll STAs according to the polling list. In this way all the STAs in the polling list will have an access to the medium in turn. In PCF the IFS that used in the same manner as DIFS in DCF, is shorter in time and is called PIFS. Because the CFP will repeat with nearly constant intervals, the PCF provides a time-bounded service for real-time traffic like video conferences.

## 4.3 PHY layer

The task of the physical layer is to send the frames received from the MAC layer to the air. The IEEE 802.11 defines three different types of physical layers: Direct Sequence Spread Spectrum (DSSS), Frequency Hopping Spread Spectrum (FHSS) and InfraRed (IR). The PHY layer can be divided into two separate parts: Physical Layer Convergence Procedure (PLCP) and Physical Medium Dependent (PMD) sublayer. Each type of PHY layers has its own PLCP and PMD sublayers.

### *FHSS PHY*

The FHSS layer supports a maximum of 2 Mbit/s data rate. The principle on 802.11 FHSS is that the channel is divided into a series of 1 MHz channels that are hopped through according to hopping sequences [IE[3]99a, pp. 177. Frequency hopping is a spread spectrum technique that has some robustness against narrow band interference. The FHSS PHY uses GFSK modulation.

### *IR PHY*

The IR PHY uses almost visible light in the range from 850 nm to 950 nm for transmission. The frequencies that the IR PHY uses don't penetrate walls. Therefore, the IR traffic doesn't cause interference to other IR systems located e.g. in other rooms. The same wavelengths are used e.g. in remote controls and IrDa equipment [IRD03]. The IR PHY is not directed so the receiver and transmitter can locate anywhere in the range of IR PHY, which is typically 10 m and at most 20 m [IE[3]99a, pp. 224]. The IR PHY supports data rates of 1 and 2 Mbit/s. The modulation used is Pulse Position Modulation (PPM). 16-PPM is used for 1 Mbit/s and 4-PPM is used for 2 Mbit/s speed.

### *DSSS PHY*

The most IEEE 802.11 compliant devices sold nowadays use the DSSS PHY layer. IEEE 802.11 defines a maximum of 2 Mbit/s transmission speed using DSSS.

#### DSSS PHY PLCP sublayer

The MAC Protocol Data Unit (MPDU), from the MAC layer, is added with PLCP preamble and header to comprise PLCP Protocol Data Unit (PPDU) (Figure 8).

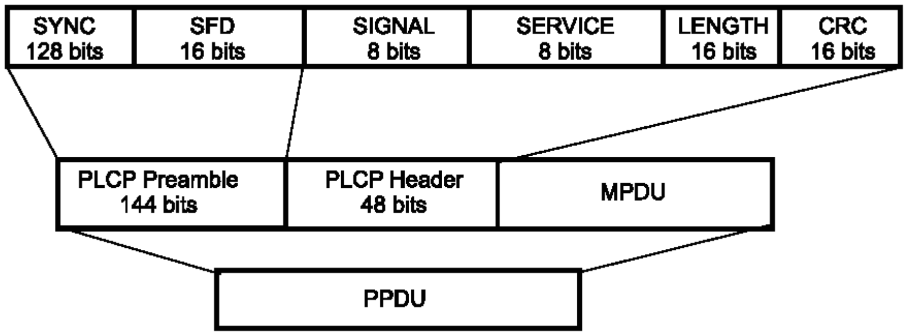


**Figure 8 DSSS PHY PLCP protocol data unit [IE[3]99a, pp. 196].**

The PLCP preamble is used to acquire the incoming signal and synchronization of the demodulator. The PLCP header contains information about the MPDU from the sending DSSS PHY. The preamble and header are sent with 1 Mbit/s but the MPDU can also use 2 Mbit/s speed. The maximum length of the MPDU is 8191 bytes [IE[3]99a, pp. 205].

### DSSS PHY PMD sublayer

The PMD sublayer takes the DSSS PHY PPDU and transmits it to the air (Figure 9). The transmitted data rate is 1 Mbit/s or 2 Mbit /s for DBPSK and DQPSK respectively.

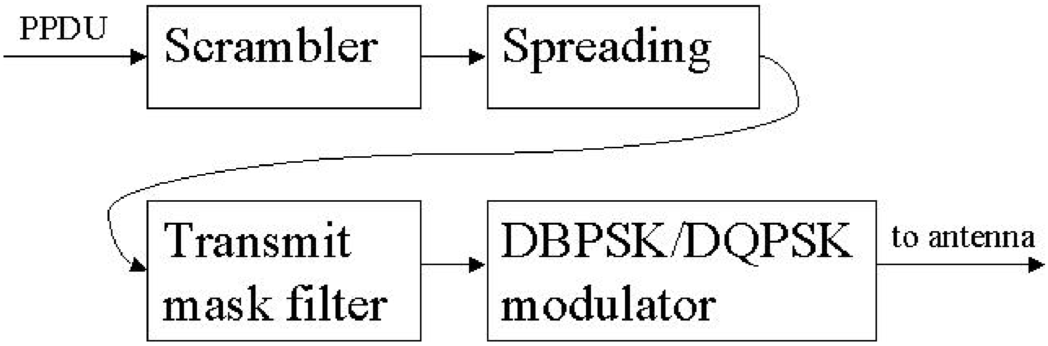


**Figure 9 DSSS PHY transmitter block diagram.**

The direct sequence transmission means that the data stream is multiplied (spread) with a chipping sequence, which has a higher rate than the data sequence. The resulting (Figure 10) signal will have higher bandwidth and lower power spectral density than the original one. The energy though remains the same. In IEEE 802.11 an 11-digit Barker sequence is used as the chipping signal. The processing gain $G_p$ is derived from the chipping rate ($R_c$) to signal rate ($R_b$) ratio

$$Gp = \frac{R_c}{R_b}. \qquad (1)$$

The processing gain in decibels describes how much improvement in Signal to Noise Ratio (SNR) can be achieved when the received signal is multiplied with the Barker sequence. The spreading process helps the transmitted signal to cloak behind background noise. This is useful if the wireless transmission is wanted to have low interception probability. The de-spreading process is the opposite (in Figure 10 the spread signal becomes the original signal) of spreading because it separates the desired signal from noise.

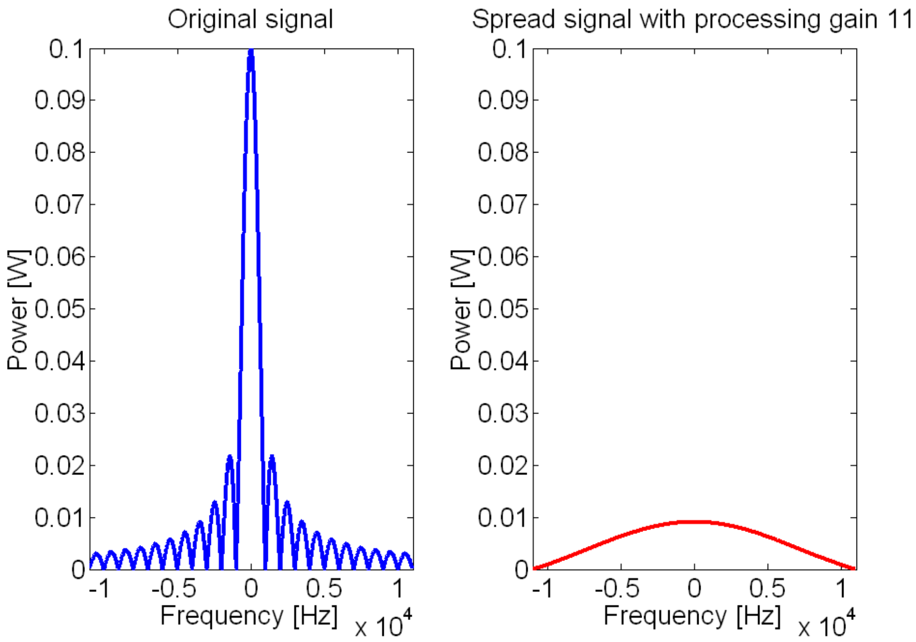


**Figure 10 Signal spreading with processing gain 11. Original signal power is 100 mW. The spread signal has a clearly reduced power and wider bandwidth.**

The energy spread of a single channel that is defined in IEEE 802.11 standard pp. 219 is shown in Figure 11. In Europe the maximum allowed isotropically radiated power is 100 mW.

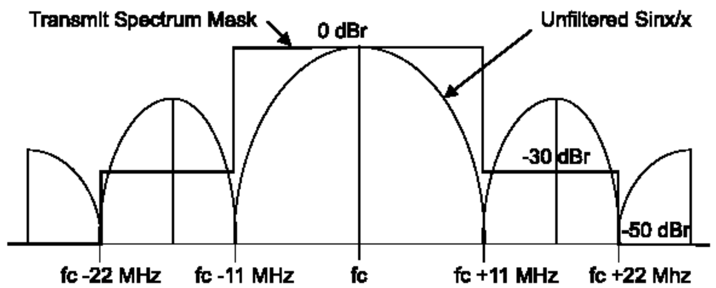


**Figure 11 The transmit spectrum mask of a single channel [IE$^3$99a, pp. 219].**

As seen in Figure 11, the single channel power is mostly confined to the 22 MHz frequency band. In IEEE 802.11, the channel spacing is 5 MHz (Table 1). The center frequencies are defined by

$$f_c(n) = \begin{cases} 2412 + (n-1) \times 5 \; [MHz], & 1 \le n \le 13, \; n \in \mathrm{N} \\ 2483 \; [MHz] & , \; n = 14 \end{cases} \tag{2}$$

where $f_c$ is the center frequency for channel number n.

Due to the transmit power spectrum shown in Figure 11, a single channel in use causes interference to the neighboring channels in a 22 MHz band around the channel center frequency. If a spacing of 25 MHz is used between deployed IEEE 802.11 channels, the interference will be small. This results in a total of three possible non-overlapping channels.

**Table 1 IEEE 802.11 channel allocations.**

| Regulatory domain | Channel numbers | Channel center frequencies [GHz] |
|---|---|---|
| US (FCC)/Canada (IC) | 1 to 11 | 2.412-2.462 |
| Europe, excluding France and Spain | 1 to 13 | 2.412-2.472 |
| France | 10 to 13 | 2.457-2.472 |
| Spain | 10 to 11 | 2.457-2.462 |
| Japan (MKK) | 14 | 2.484 |

### 4.4 Mobility in IEEE 802.11

The two types of networks, infrastructure and ad-hoc network, both provide the STAs a possibility to move around in the network's coverage area. In infrastructure networks the STA is always associated with no more than one AP. The STA can change the AP it is associated with (this is called handover), but the standard doesn't define how an STA should choose the AP if there are several alternatives. For a handover to happen as seamlessly as possible, the APs have to be in the same distribution system and in the same ESS. After the STA has associated with a new AP, the old and the new AP would have to exchange information between them in order to make the reassociation as seamless as possible. The information exchanges between APs, however, are not defined in the standard, so in practice the exchange of BSSs is best handled by APs of the same manufacturer. If an STA performs a handover between APs that don't belong to the same ESS or don't constitute an ESS, some user data may be lost, because IEEE 802.11 doesn't provide support for user mobility in that case.

## *5 Approved supplementary IEEE 802.11 standards*

IEEE 802.11 provides a connection with a maximum speed of 2 Mbit/s. IEEE 802.11a and b were developed in 1999 and they make data rates of 54 and 11 Mbit/s possible respectively. The a, b and d versions are mainly PHY layer extensions to IEEE 802.11. Nowadays the IEEE 802.11b is the most widely used WLAN standard. The 802.11a uses frequencies in the 5 GHz band and IEEE 802.11b frequencies in the 2.4 GHz band. In Europe the 802.11a is not widely used because the frequency band in the 5 GHz is reserved for the European WLAN standard Hiperlan/2.

### 5.1 IEEE 802.11a

The 802.11a uses Orthogonal Frequency Division Multiplexing (OFDM) in the 5 GHz band. Because the 802.11a uses higher frequency than 802.11b its signal attenuates more rapidly. In the United States 12 channels are defined in the 5 GHz band. Each channel in 802.11a is 20 MHz wide and it consists of 52 sub-carriers.

The advantage of OFDM is that the transmission bandwidth is large enough, so that possible interference sources don't affect all the sub-carriers. OFDM is also more robust against Inter Symbol Interference (ISI) caused by multipath propagation than the DSSS technique. If the transmission channel suffers from e.g. narrowband interference, OFDM makes it possible to use better coding or more robust modulation method to combat the interference. Those methods of course lower the maximum possible data rate or increase the overhead of transmission. The modulation and coding options can be see in Table 2. The required operation modes are 6 12 and 24 Mbit/s speeds [IE[3]99b, pp. 3].

**Table 2 IEEE 802.11a modes [IE[3]99b, pp. 9].**

| Speed, Mbit/s | Modulation and coding rate, R | Coded bits per sub-carrier | Coded bits per symbol | Data bits per symbol |
|---|---|---|---|---|
| 6 | BPSK, ½ | 1 | 48 | 24 |
| 9 | BPSK, ¾ | 1 | 48 | 36 |
| 12 | QPSK, ½ | 2 | 96 | 48 |
| 18 | QPSK, ¾ | 2 | 96 | 72 |
| 24 | 16-QAM, ½ | 4 | 192 | 96 |
| 36 | 16-QAM, ¾ | 4 | 192 | 144 |
| 48 | 64-QAM, 2/3 | 6 | 288 | 192 |
| 54 | 64-QAM, ¾ | 6 | 288 | 216 |

## 5.2 IEEE 802.11b

IEEE 802.11b was approved in 1999 and it is a supplement to IEEE 802.11 [IE[3]99c]. The main difference to IEEE 802.11 is that 802.11b uses higher transmission rates. IEEE 802.11b uses Complementary Code Keying (CCK) [IE[3]99b, pp. 43] to accomplish 11 Mbit/s transmission speed. An optional method to achieve 11 Mbit/s speed, called Packet Binary Convolutional Coding (PBCC), is defined in [IE[3]99b, pp. 45]. Other operation speeds for 802.11b are 1, 2 and 5.5 Mbit/s. The 802.11b equipment are backward compatible with 802.11.

In the PHY layer of IEEE 802.11b there are two preamble possibilities, short (this is an optional part of the standard) and long, instead of only one defined in IEEE 802.11. The short preamble is used if throughput efficiency is important. Also in voice communications, the short preamble can be better than the long one, since it reduces overhead of the transmission and therefore delays are shorter. The long preamble is like in Figure 8. The short preamble differs from the long one in the length of the SYNC field in Figure 8. The short preamble's SYNC field is only 56 bits long and therefore the entire preamble is only 72 bits long. The PPDU using a long preamble is transmitted as defined in Section 1.3. The PPDU using a short preamble uses 1 Mbit/s transmission speed in the PLCP preamble and 2 Mbit/s in the PLCP header. The transmission speed of the MPDU can't be 1 Mbit/s, when short PLCP preamble is used. Other 802.11b speeds are supported, though.

## 5.3 IEEE 802.11d

IEEE 802.11d standard was approved in 2001 [[IE[3]99d]]. The standard is supplementary to the MAC layer of 802.11 and the existing standards of IEEE 802.11a and b. The goal of the standard is to enable wider use of Wi-Fi equipment outside the United States. The standard defines how APs communicate with STAs to share information about allowed frequency channels and transmitter powers. Devices conforming to this standard don't have to be country specific anymore because they can dynamically configure themselves.

### 5.4 IEEE 802.11f

This supplement to IEEE 802.11 defines the communication between APs in the same distribution system [IE$^3$99e]. The Inter Access Point Protocol (IAPP) is used for communicating between APs. The communication between APs will enable the STAs e.g. improved mobility between BSSs.

### 5.5 IEEE 802.11g

This standard defines Extended Rate PHY (ERP) layer to IEEE 802.11. The channels available for IEEE 802.11g are the same as for IEEE 802.11, so there are only three non-overlapping channels. The high transmission rates are made possible by the OFDM technique. The standard defines mandatory transmission and reception rates of 1, 2, 5.5, 11, 6, 12, and 24 Mbit/s from which only the three highest use OFDM and the others are already defined in IEEE 802.11b. Like IEEE 802.11a also the g has a maximum rate of 54 Mbit/s. The g standard is backward compatible with IEEE 802.11b, which means that the equipment conforming to IEEE 802.11g have to be able to communicate with IEEE 802.11b equipment. However the 802.11b devices can't "hear" the 802.11g devices and the 802.11b devices appear to g devices as noise. To combat this interoperability problem the 802.11g standard defines that a BSS that has both 802.11g and 802.11b compliant devices should have some kind of protection mechanism [IE$^3$99f, pp. 9] (e.g. RTS/CTS). The use of e.g. RTS/CTS reduces the throughput of the network.

## *6 VoIP over WLAN*

VoIP over WLAN means provision of a voice transmission over a wireless link of a local area network. Differently from cellular systems like GSM for example the IEEE 802.11 family standards were initially not intended to provide stringent quality requirements. As such they are not most suitable for real time voice communication. However, given sufficient amount of transmission capacity the 802.11 compliant systems can also be used for voice communication.

In this laboratory work we measure the quality of a voice communication in different radio environment. We study how the attenuation in the channel and the packet loss impact total end to end voice quality. The measurements illustrate the applicability of WLAN for VoIP.

In this work we investigate the connection quality during one voice call. Such approach overlooks the issues related to the connection establishment times and problems related to multiple access.

## *7 Quality of service in voice communication*

The connection quality depends on multiple equipment and communication environment related factors. The main end-to-end quality assignment is done by human beings. However, it is often not practical to make subjective tests. The human perception is a psychological measure and as such it is difficult to define quantitatively. The approach taken by technical community is to map the technically measurable connection impairments to human perception values. In simplified form one can understand it as mapping: a packet loss Y corresponds to perception quality level X.

In communication systems the voice quality is degraded at each internal processing stage. The system internal factors are related to the signal processing in the equipments and the signal transmission environment. For example quality is degraded due to the voice quantization, compression, delays, bit errors e.t.c.

In order to quantify the total degradation ITU-T recommendation G.107 suggest using so called E-model (Impairment factor method). Accordingly to this methods the individual impairments are assumed to be additive. Individual degradations are assigned values describing their contribution to the final end-to-end quality. The total degradation is a sum of individual degradations.

In this laboratory work we concentrate mainly on the radio channel impact to the total end-to-end quality. We are not measuring impairment in the individual units but change the channel parameters and observe the change impact to the end-to-end voice quality. The final quality is assigned by the tester who listens the received signal.

## 7.1 QOS metrics

The end-to-end quality is a human reception based parameter. The main metrics describing it are mean opinion score (MOS), percentage good of better (%GoP), and percentage poor of worse (%PoW). Thos metrics are described in the appendix B of ITU-T recommendation G107 [G.107]

These metrics are opinion based and need humans for assigning them. Technically more easier is to remove the human tester. One possible to approach is simply to compare the received and transmitted signals. The possible differences are mapped to the opinion scores. For example such approach is taken in assigning Perceptual Speech Quality Measure (PSQM), outlined in ITU-T recommendation P.861. Unfortunately the speech encoder quality measure does not encounter for network impacts. It does not have a way do describe the delays, jitters, packet drops and other phenomena occurring in the network. Because of that PSQM is not very suitable for describing VoIP connection quality.

In this laboratory work we use a simplified method for assigning Mean Opinion Score.

### 7.1.1 Mean Opinion Score

MOS is a subjective metric that describes the quality of voice perception by human. MOS is expressed in the scale from 1 to 5 where 1 is the lowest quality. A general voice quality test procedure with the example sentences for test is given in [P.85]. The considerations related to testing of a voice quality in telecommunication network are given in the ITU-T recommendation P.800 [P.800]. Because the MOS tests are involving the human listeners they are most suitable for assessing communication system quality. However, the testing process is time consuming and expensive.

The MOS test is subjective test. The test person listens a group of sentences and gives the personal opinion about the received voice quality. MOS ratings are given in the table 5.1 below.

Table 5.1: The Mean Opinion Score (MOS) rating. (Table B.2/ITU-T P.800)

| Rating | Descriptions |
|---|---|
| 1 | Very annoying. No meaning understood with any feasible effort |
| 2 | Annoying. Considerable effort required |
| 3 | Slightly annoying. Moderate effort required |
| 4 | Perceptible but annoying. Attention necessary; no appreciable effort required |
| 5 | Imperceptible errors. Complete relaxation possible; no effort required. |

# 8 *Quality of service in packet network*

Most common parameters describing packet network connection are: End-to-end delay, Packet Jitter, Packet Loss, Echo, Speech and Noise levels.
In this laboratory work we look at the first three of those.

## 8.1 End-to-end delay

The end to end delay, called also latency, is time the voice takes to pass through the whole communication system. The end-to-end delay is sum of processing, queuing, transmission and propagation delays.

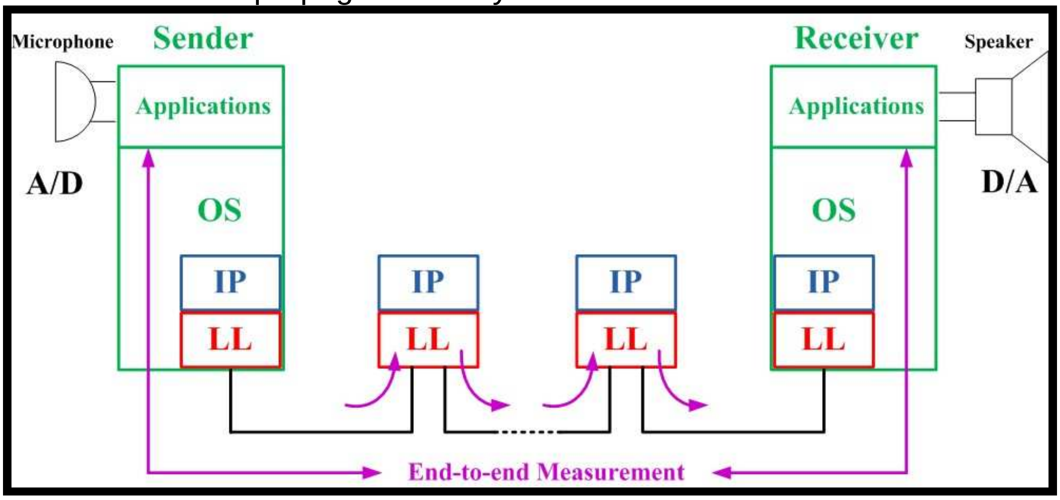


Figure 6.1: End-to-end measurement delay

In a VoIP the voice connection is established over IP network. The delay in such network is related to converting the analogue voice into digital packets and transmission of those packets over IP network. The voice conversion contains sampling and packet creation; serialization or putting packets into transmission interface. In the transmission system the packet has delays due to the processing in different network nodes (packetization delay), waiting due to the queue system and in the transmission environment.
Depending on the encoding scheme the packet is usually created over samples in 10 -50 ms. The packet is encode and decoded which process adds additional 10 -50 ms. The serialization and packet processing are taking time of few milliseconds. The queuing delay

depends on network congestion factor. In uncongested network it takes time of order of 5-200 ms. In wireless links the distances are relative small. However if to consider the whole propagation path containing also all wired links the delays reaches easily to be 5 – 120 ms.
The voice quality is impacted by the accumulative delay in the network. The relation between the end-to-end delay and voice quality is shown below

Table 6.1: End-to-end delay impact to voice Quality

| End-to-end delay (ms) | Voice quality |
|---|---|
| lower than 150 | Good |
| between 150 and 400 | Acceptable |
| higher than 400 | Poor |

The impact of delay should not be considered only in one direction. In computer system with alone standing microphone and loudspeakers the received voice is easily picked up by the microphone and send back to the source. In order to avoid such annoying echo the system has to include an echo canceller. A general rule is the echo canceller is a necessity for Round trip delays more than less than 25.

## 8.2 Jitter in Packet Voice Network

A processing delay in packet network nodes and in the computer operating systems are usually not well controlled. The uncontrolled delays lead to variation in packet arrival times – jitter. The jitter can be measured in various ways. Most common is the IETF definition of jitter: mean deviation of received packet spacing (reception times) compared to the sender packet spacing.
One can assume that the transmitted packets have uniform spacing. The variation observed at the receiver side is due to the congestions, processing delays, improper queuing.

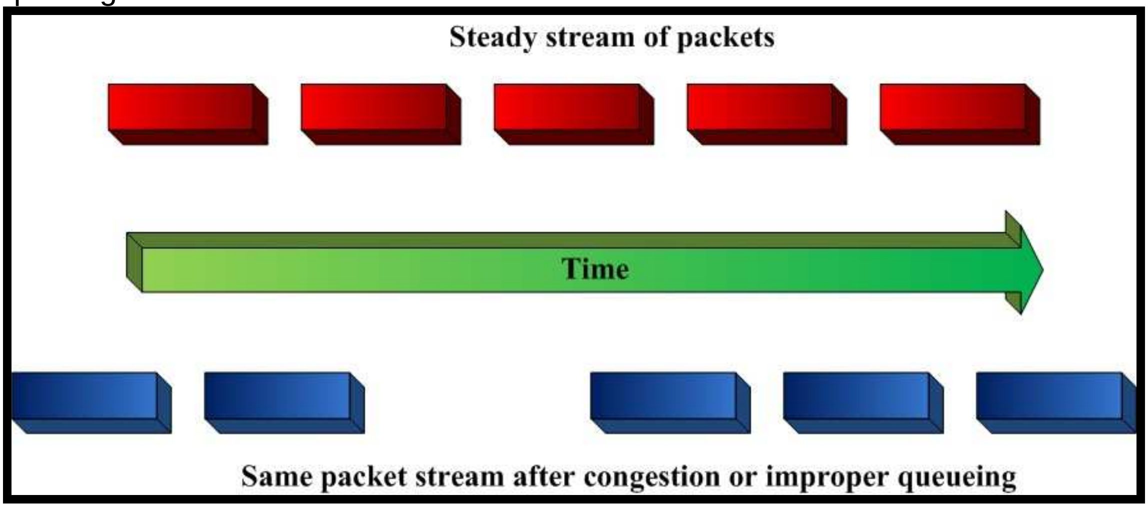


Figure 6.2: The illustration of a steady stream of packets is handled.

The differences in arrival variations can be smoothed out by a play-out delay buffer. The play-out buffer gathers the received packets into a buffer and reads them out with constant frequency. The time difference between the packets and its variation is estimated from the time stamps included in the Real-Time Protocol (RTP). Sometimes the play-out delay buffer is referred to as the de-jitter buffer.

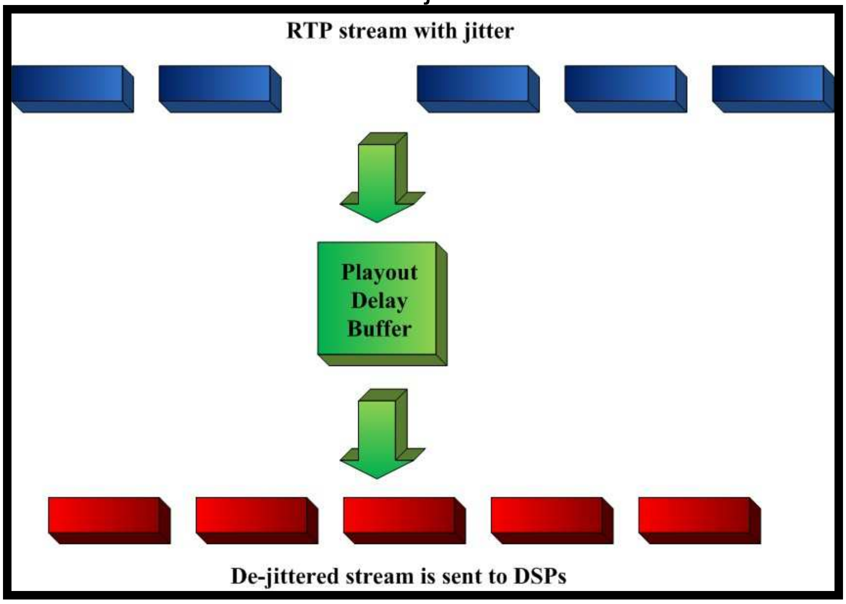


Figure 6.3: The illustration of the jitter is handled.

In order to comply with the total delay requirement the large jitter can not be compensated. The packets arriving too late are discarded and the dropouts are heard in the audio. In order to avoid clicks in the output audio the advanced VoIP systems compensate the missing signal by interpolating it.

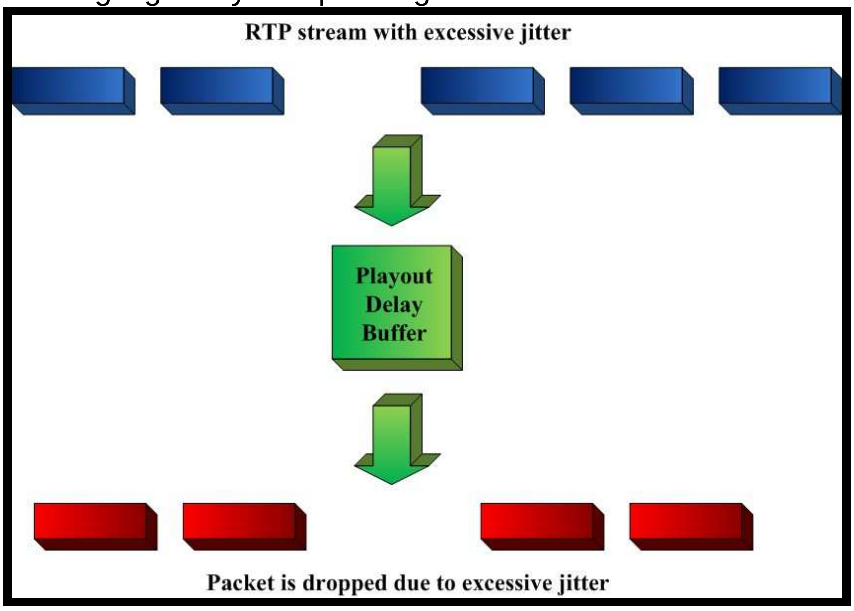


Figure 6.4: The illustration of the excessive jitter is handled.

## 8.3 Packet Loss

Usually the VoIP application does not have time for retransmissions and therefore uses UDP protocol. During the bad connections or congestions the UDP packets are lost. The ability of the VoIP application to cope with the packet loss depends on the used coding/decoding methods. There exist methods that allow as much as 50% of packet loss rate.

Table 6.2: Packets loss impact on voice quality

| Percentage of losses | Voice quality |
|---|---|
| lower than 5 % | Good |
| higher than 5 % | Poor |

Beside the total packet loss also the distribution of the lost packets is crucial. If many neighboring packets are lost missing voice piece can not be compensated and the perceived voice quality is significantly degraded.
A commonly used PCM encoded can cope with 1% of lost packages without the loss concealment techniques and up to 10 % of lost packages with concealment techniques.

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# APPENDIX A

## ITU-T recommendation P.85 Annex A

This annex gives examples of messages for testing MOS.

Two applications were involved in this experiment: mail order shopping (M) and railway traffic information (R). Three
messages are given for each application.
M1: Miss Robert, the running shoes colour: white, size: 11, reference: 501-97-52, price: 319 francs, will be delivered to you in 1 week.
M2: Mr. Johnson, the multistandard TV set with remote control, 36 cm screen, reference: 811-61-32, price: 2 492 francs, will be delivered to you in 3 weeks.
M3: Mr. Moore, the electric drill D162, power: 550 watts, 2 speeds, reference: 481-20-30, price: 499 francs, will be delivered to you in 2 weeks.
R1: The train number 9783 from Glasgow will arrive at 9:24, platform number 3, track G.
R2: The train number 7826 to Ipswich will leave at 12:20, platform number 9, track A.
R3: The train number 4320 from Birmingham will arrive at 5:44, platform 2, track C.