Laboratory Course in Communications Engineering 2

Interference measurements in DVB-T system

Background material and exercises

Table of Contents

Abbreviations

ACE	Active Constellation Extension
ACI	Adjacent Channel Interference
ACLR	Adjacent Channel Leakage Ratio
ACS	Adjacent Channel Selectivity
APS	Amplitude and Phase Shift keying
BER	Bit Error Ratio
BCH	Bose-Chaudhuri-Hocquenghem code
C/N	Carrier-to-Noise
DVB	Digital Video Broadcasting
DVB-C	Digital Video Broadcasting - Cable
DVB-H	Digital Video Broadcasting - Handheld
DVB-S	Digital Video Broadcasting - Satellite
DVB-T	Digital Video Broadcasting - Terrestrial
ELG	European Launching Group
ETSI	European Telecommunications Standards Institute
FEC	Forward Error Correction
FER	Frame Error Ratio
FFT	Fast Fourier Transform
HD	High Definition
ICI	Inter Carrier Interference
LDPC	Low Density Parity Check
MER	Modulation-to-Error Ratio
MFN	Multi-Frequency Network
MoU	Memorandum of Understanding
MPEG	Moving Pictures Experts Group
MPEG TS	Moving Pictures Experts Group Transport Stream
OFDM	Orthogonal Frequency Division Multiplexing
PAPR	Peak-to-Average Power Ratio
QAM	Quadrature Amplitude Modulation
QEF	Quasi Error Free
QPSK	Quadrature Phase Shift keying
RS	Reed Solomon
RWB	Resolution Bandwidth
SFN	Single-Frequency Network
S/I	Signal-to-Interference
SIR	Signal-to-Interference-Ratio
SINR	Signal-to-Interference-plus-Noise-Ratio
SNR	Signal-to-Noise-Ratio
STB	Set Top Box

TPS	Transmission Parameter Signalling
UHF	Ultra High Frequency
VHF	Very High Frequency

1 Introduction

Transition from analogue to digital television began in the 1990s, as several TV- and radio-companies, consumer electronics manufacturers and regulatory bodies' came together to discuss the formation of a group that would oversee the development of digital television. The group, which was formed during years 1991-1993, expanded to include the major European media interest groups, the consumer electronics manufacturers, common carriers and regulators. The group vas called European Launching Group (ELG) and they established common rules, with which everyone should agree when developing digital television. The agreement is called Memorandum of Understanding (MoU) and it obligates its members to appreciate their common requirements and agendas. According to the agreement, the members must establish trust and mutual respect. The MoU was signed by all ELG participants in September 1993, and the Launching Group renamed itself as the Digital Video Broadcasting Project (DVB). [1]

The analogue transmission technology was well quite inefficient. Good reception required high SINR level and lot of spectrum transmitting one TV channel. Digital technology uses radio frequencies more efficiently; it takes advantage of channel coding and data compression. In the same spectrum area where analogue TV could transmit only on TV channel a digital system could multiplex multiple channels. The channel coding allows receiving those channels at much lower SINR level and can use signal processing methods for combating poor radio channel conditions. That is why digital technology improves the quality of video and audio especially in areas, which has had problems with receiving analogue TV-transmission because of for example environment obstacles. With digital technology many disturbances, like static and ghost image, have disappeared. [2]

The digital TV has become a qualitative change in the spectrum utilization. For transmitting same amount of TV channels we can use less power and spectrum. The next step in increasing the spectrum utilization is to reuse the spectrum more densely. The spectrum reuse is limited by the interference generated among the transmitters. The digital systems can sustain certain amount of interference but if the interference level is too high the receiver is unable to correct the errors. It in analogue system the interference increase resulted a relatively gradual decrease of reception quality in digital system the change of the reception quality is more abrupt. As long as the receiver is able to decode the data the reception is relatively error free. If the disturbance threshold is exceeded the most of the bits will be erroneous and the reception is nearly impossible. The errors in digital systems manifest itself as pixelisation, freezing of picture and disturbances in sound. Also with high resolutions can appear MPEG-disturbances (Moving Pictures Experts Group), which originate from limited bit rate and data compression. In that case the resolution of the transmitted video could be too low and it becomes apparent with blurry spots and digital noise. For the high resolutions, there are HD-technologies (High Definition) and second generation DVB-standards, like DVB-T2.

1.1 Aims of the work

In this laboratory work you learn the basics features of DVB-T physical layer operations. You will learn the structure of the OFDM signal and you understand the basics principles of coding. You will know how a DVB-T system is parameterized and how different options sustain disturbance from other transmitters.

In the laboratory work you use a DVB-T system for investigating the impact of the interference on the system performance. After the laboratory works you will know what the interference is how to classify and characterize the interference and how different types of interferences impact the receiver performance.

After this laboratory you will be able to construct a measurement system for measuring co-channel and adjacent channel interference. You know how the deployment of strong codes impacts the performance of receiver and how the receiver has to be protected.

2 Signal quality measures

A communication link quality is described by the ratio of the useful signal power to the interfering signal power. The interference is any kind of disturbance of the useful communication signal. The disturbance can be due to the impact of the transmission environment, due to the imperfections in transmitter and receiver, or due to signals emitted by other transmitters.

Signal-to-Interference Ratio, Signal-to-Noise Ratio (SNR), Carrier-to-Interference Ratio Carrier-to-Noise Ratio (CNR) and Modulation Error Ratio (MER) are closely related to each other and in some cases they result the same value. These might be hard to distinguish from each other and that is why these terms have been shortly handled.

Signal-to-Interference Ratio (SIR) average modulated carrier power ratio to the average received co-channel interference power. It is described also as cross-talk between the useful and other transmitters.

Carrier-to-Interference Ratio (CIR) is the average carrier power ratio to the average received co-channel interference power. The CIR is evaluated at the antenna input while the SIR is evaluated after the modulation. In conventional TDMA and FDMA receivers CIR and SIR are the same. In CDMA the CIR is evaluated before spreading operation and SIR after the spreading.

Signal-to-Noise-Ratio (SNR) is the ratio of the average useful signal power to the background noise power.

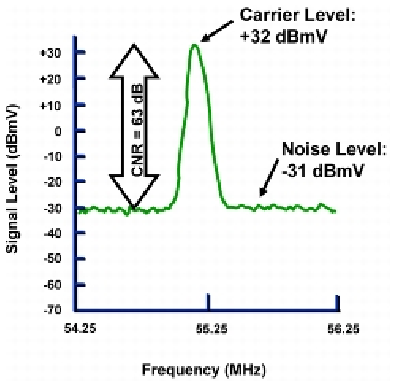
SNR is a common term, and it can refer to both radio frequency and baseband measurements. SNR has generalized to be used in baseband measurements however. CNR is often used in radio frequency measurements. For example, CNR refers to signal to noise ratio given by spectrum analyzer. CNR is gotten by calculating the difference between the signal power and noise power levels. Examples of CNR and SNR measurements can be seen from figure 7.

Signal-to-Interference-plus-Noise-Ratio (SINR) describes the total ratio of the average signal power to the average interference and noise poser.

Modulation Error Ratio (MER) is defined as the ratio between the average received symbol power and the average modulation error power. It describes the error of the constellation in digital modulation. Symbol in this context means one modulated carrier. Modulation error is gotten by calculating the difference between the real value of the received symbol and the value of target symbol (Fig. 8) [10].

Bit Error Ratio (BER) number of correctly received bits divided by total number of bits.

The BER can be measured by creating a bit stream that is known for both transmitter and receiver. The BER is computed by comparing the received bits to the known bits in the receiver. Such approach works very well in simulations. Unfortunately in a real system the receiver might not be able to recognize a full frame after that lose the pointer in the bit stream, ie. which bits ate transmitted. In order to avoid such problems the BER is usually computed by comparing the received bits to the bits derived after the decoding the received data. The correctness of the decoded data frame is validated by checking its CRC. The correct decoded bits are encoded again and the "code corrected" encoded bits are compared to the bits in the receiver input.



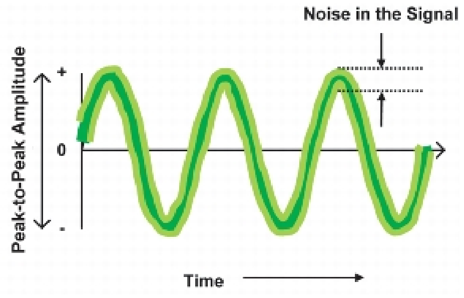


Figure 2-1 Radio frequency CNR measurement on the left, baseband SNR measurement on the right.

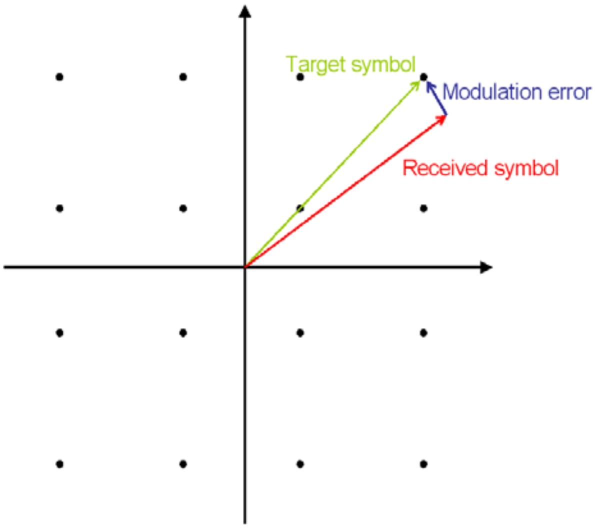


Figure 2-2. Constellation of the 16QAM-modulation, modulation error

Frame Error Ratio (FER) number of correctly received frames divided by total number of frames.

During these laboratory measurements, CNR of the DVB-signal can be measured with the spectrum analyzer and MER can be measured by using DBA-program.

2.1 Interference

In this work we investigate interferences generated by other spectrum users: co-channel interference and adjacent channel interference.

The received signal quality is usually described by SINR value. In case the other user interference is much higher than the noise level the system quality can be described solely by interference level (CIR). Such situation is called interference limited system. The CIR model is suitable for instance for modeling a cellular system with low reuse factor and high TX powers. In such case the disturbance from transmitters in other cells is much higher than the noise power and the latter can be ignored.

In case the interference level is much lower than noise level the system is called noise limited system. An example of a noise limited system is a cellular system with extremely large cells.

2.1.1 Co-channel interference

Co-channel interference is the disturbance generated by transmitters using the same frequency.

Due to the attenuation the frequency can be reused at some distance. The signals received from the frequency reusing transmitters are the co-channel interference. The amount of co-channel interference has to be limited such that our useful signal can be received with required quality. Usually the co-channel transmitters are deployed such that the co-channel interference does not violate the SINR requirement at the receiver. That can be achieved by using low enough transmission power or by increasing frequency reuse distance.

The violation of the co-channel interference limit could occur for instance: if the network is planned to be too tense (i.e. poor frequency planning), if the attenuation changes due to the changing weather conditions etc.

2.1.2 Adjacent channel interference

Adjacent Channel Interference (ACI) is the average signal power (interference) generated by the transmitters using adjacent channel. Adjacent channel interference consist two parts leakage power from the adjacent channel transmitters and the adjacent signal power the receiver is not able to reject in the receiver filters.

The leakage power is described by Adjacent Channel Leakage Ratio (ADLR). This is interference due to the fact that the transmitters are not able to confine the transmitted signal into the allocated channels. Some of the power is spilled out to the neighboring channel. How much power a transmitter is allowed to generate on the neighboring channels is specified by the spectrum emission mask. Such masks are the requirements set by frequency allocation officials.

Adjacent Channel Selectivity (ACS) describes how much the receiver filter is suppressing signal in the neighboring channels. The receiver filters are not able to remove full signal in the rejection band they only suppress the signal. How much the receiver filter has to suppress the signal in neighboring channels is given by system specifications and usually is standardized. For instance the Requirements for Nordic Digital TV receivers “NorDig Unified Requirements” specifies that ACS in adjacent channel should be at least 28 dB and on other channel (n+2 or more) 38 dB.

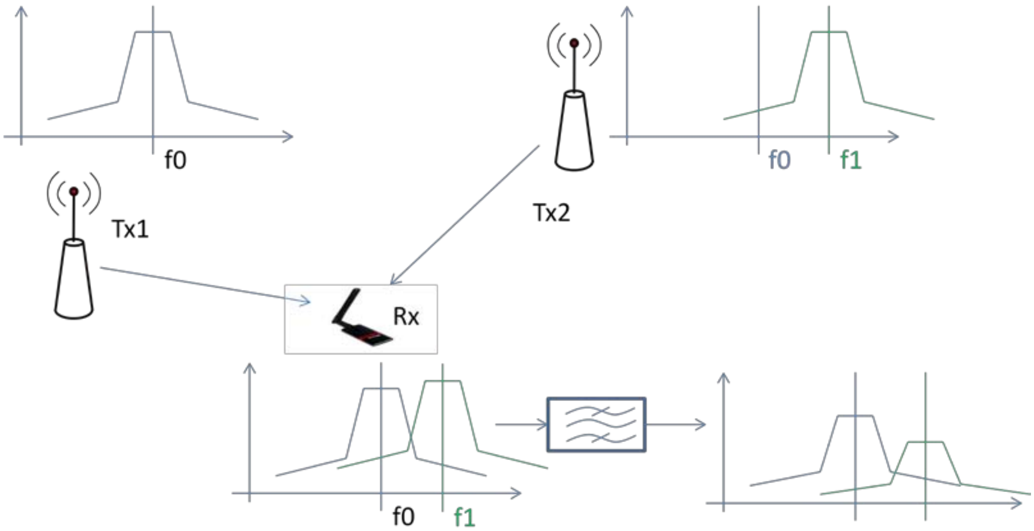


Figure 2-3 Illustration of the adjacent channel interference. Our transmitter is using frequency f0, adjacent channel is on frequency f1. The adjacent channel transmitter Tx2 emits some power in channel f0 and our receiver receives it. Our receiver suppresses the adjacent channel by the amount of ACS.

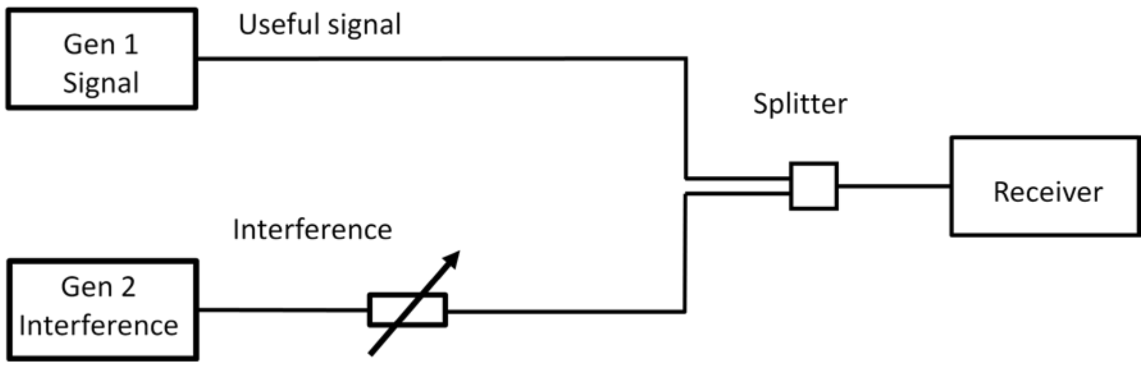


Figure 2-4. General setup for the interference measurement.

2.1.3 Interference measurements

The interference level is expressed by its ratio to the useful signal level (by SIR or SINR). How much interference is acceptable at the receiver is usually fixed for certain BER level. It describes of what is the CIR level where the target BER is still satisfied. The acceptable interference level gives a threshold for SINR that the interference level should satisfy.

At certain SNR level the interference can be described also as the amount of interference increase required for degrading connection quality between initial and final quality levels. For instance at certain SNR how much interference has to be increased in order to degrade the BER from 10^-6 to 10^5.

The interference impact is measured by summing the interfering signal to the useful signal and measuring the received signal quality. A schematic of the measurement setup is given in Figure 4.

3 Digital Video Broadcasting Project

The Digital Video Broadcasting Project is an industry-led consortium committed to designing open technical standards for the global delivery of digital television and data services. DVB began as a European project, but currently it has spread globally to every continent.

European Telecommunications Standards Institute (ETSI) has published multiple standards with respect to DVB, some of which have been presented in the table 1. Nowadays there are over 500 million DVB-receivers in use in the world, which of at least 140 million are DVB-S-receivers and 150 million are DVB-T-receivers. DVB-S/S2-standard forms the basis of satellite-TV all over the world, DVB-C is the most used cable-TV-system in the world. The use of DVB-T has as well increased considerably recently in Europe and parts of Asia, Africa and Latin America. Likewise many other are planning the introduction of DVB-T. [1]

Table 1: DVB-standards

Technique	**Standard**	**Description**
DVB-C	EN 300 429	Definitions of the transmission parameters in cable systems.
DVB-C2	ETSI EN 302 769	Definitions of the transmission parameters in second generation cable systems.
DVB-H	ETSI EN 302 304	Transmission systems of the

		handheld receivers
DVB-S	EN 300 421	Definitions of the transmission parameters in satellite systems.
DVB-T	ETSI EN 300 744	Definitions of the transmission parameters in terrestrial systems.
DVB-T2	ETSI EN 302 755	Definitions of the transmission parameters in second generation terrestrial systems.

3.1 DVB-T

DVB-T (Digital Video Broadcasting – Terrestrial) is a digital transmission system, which uses terrestrial network a.k.a. antenna network for its transmissions. DVB-T uses both VHF- and UHF-frequencies. This standard was published year 1997. The first DVB-T-transmissions were sent in the United Kingdom year 1998. In several countries the transition from analogue to digital TV took place in the turn of the 2000s and the transmission standard chose was more often than not DVB-T. Today it is the most used terrestrial digital TV transmission standard. In Finland, the transition to digital TV happened in stages. Digital adapters, also known as digiboxes, came for sale in November year 2001 and 1.9.2007 the analog terrestrial TV-transmissions were shut down entirely. Then the digital terrestrial reception area covered 99,9 % of the households in Finland. [2]

DVB-T uses MPEG TS (MPET transport stream) for compressing the video, audio and data. Even though MPET is known as a video compression standard, MPEG TS is able to compress other data as well. Data to be transmitted is divided to constant length packages, which are inserted to MPEG TS multiplexer. The length of the MPEG TS multiplex packet is 188 bytes, which contains one synchronization byte. After this, the packet moves to outer coding and outer interleaving, from which it moves to inner coding and inner interleaving. The symbols, which results from there, are arranged in to frames. The OFDM-modulation, which is used in the standard, frame contains 68 OFDM-symbols and super-frame consists of 4 frames.

DVB-T is able to use hierarchical transmission, with which one DVB-T-signal is able to simultaneously transmit two different broadcasts with unequal priorities. This means sending data simultaneously to two different receivers, for example TV-broadcast to TV-receiver and data to handheld device. If the transmission route is poor, the high priority broadcast is secured at the expense of the low priority broadcast.

DVB-T uses coded orthogonal frequency-division multiplexing (COFDM). OFDM-signal consists of multiple sub-carriers, which transmit data simultaneously. While the data transmission rate of a single sub-carrier is quite low, the data transmission rate of the whole system is high because of high quantity of the sub-carriers. The sub-carriers are equally spaced and the frequencies of them are chosen so, that they are orthogonally to each other. This means that the inter carrier interference (ICI) between them is negligible and they don’t induce interference to each other. Compared to plain frequency-division multiplexing (FDM), the spacing between sub-carriers is quite wide in OFDM, which results in long symbol length. Orthogonality is executed by setting the sub-carriers to each others zero points. The sub-carriers are sinc-pulse shaped, as seen from figure 1. This means, that OFDM-is very sensitive to frequency shift. OFDM-signal spectrum is composed when several sinc-pulses are positioned next to each other as seen from figures 2 and 3. Spectrum is quite flat under the channel and resembles white noise.

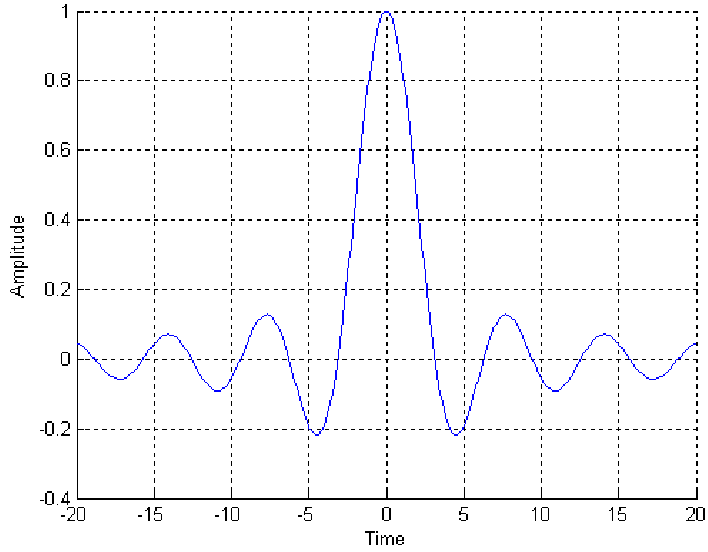


Figure 3-1. Sinc-pulse

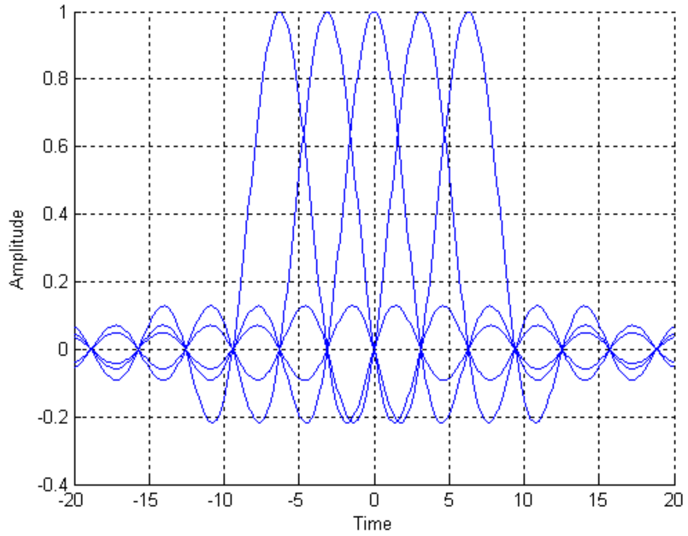


Figure 3-2. Sinc-pulses located orthogonally.

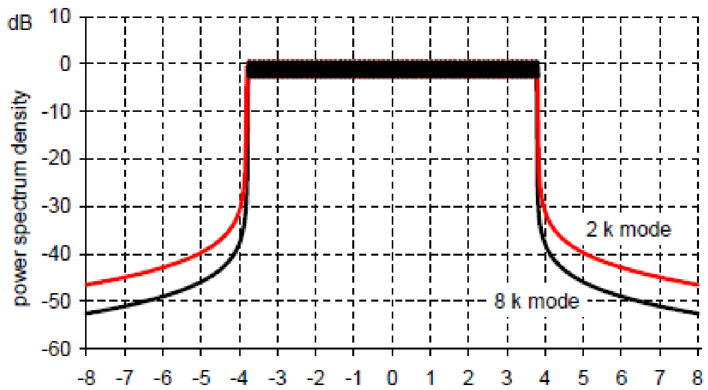


Figure 3-3. Spectrum of the DVB-signal [3]

Each of the parallel sub-carriers are modulated with data and they are combined in the frequency-domain to OFDM-symbols. Each of the sub-carriers can be modulated independently with QPSK-, 16QAM- or 64QAM-modulations. Thus the sub-carriers of the channel can be divided to several sub-channels, with which different data streams can be sent. The amount of the sub-carriers carrying the data streams can be changed if necessary when one streams needs more data transfer rate and one doesn't need that much. [4]

The symbols are transformed to time-domain with discrete inverse Fourier-transform. This results in part of the digital OFDM-signal. After that, part of the signals ending is copied to the beginning. This part is the guard interval, due to which OFDM is very resilient to problems resulted from multipath propagation. Each symbol is divided to multiple sub-carriers and during the guard interval, which is specified in the receiver, the reveived reflections of the signal do not interfere with it, but amplify the signal. This is how reflections originated from multipath propagation can be avoided efficiently. This is also why DVB-T is able to use single-frequency networks (SFN), which is a transmission network, where several transmitters send the same signal at the same frequency. Now the signals received from several sources can be combined contructively in the receiver. DVB-T standard supports guard intervals of 1/32, 1/16, 1/8 and 1/4 of the original symbol length. [3]

In DVB-T the transmission modes used are 2k- and 8k-modes. The names of the modes refer to the amount of the used sub-carriers. In the 2k-mode the amount of the used sub-carriers is 1705 and in 8k-mode it is 6817. During the modulation and demodulation, discrete Fourier-transform is used, which in practice means efficient Fast Fourier Transform-algorithm. FFT-algorithm works best, when the amount of the samples used is a 2's power. This is from which the modes have gotten their names, as in 2k-mode the possible sub-carrier number is 2^11 = 2048, from which unused are 2048 - 1705 = 343 sub-carriers. In 8k-mode the amount of the possible sub-carriers is 8192, from which unused are 1375.

In DVB-T, one channels bandwidth is used 6, 7 or 8 MHz. Usually 7 MHz bandwidth is used in VHF-band and 8 MHz bandwidth in UHF-band. The spacing between sub-carriers are 4464 Hz in 2k-mode and 1116 Hz in 8k-mode, when the used bandwidth is 8 Mhz. Then with both modes the used bandwidth is 7,61 MHz. If also the unused sub-carriers would be used, the bandwidth would be over 9,1 MHz. [3]

For error correction there is both inner and outer coding in the standard. Inner coding used in DVB-T standard is punctured convolutional coding, which is based on mother convolutional coding with code rate 1/2 and 64 states. Convolutional coding is often denoted as forward error correction (FEC). The coding rate of the FEC-coding determines how much of the bit stream is used for transmission of the useful data and how much is used for error correction. For example, when the code rate is 2/3, useful data constitutes 2/3 of the bit stream and error correction constitutes 1/3 of the bit stream. The code rates defined in DVB-T standard are 1/2, 2/3, 3/4, 5/6 and 7/8. [3]

The polynomials used in DVB-T's preconvolutional coding are $G_1 = 171_{OCT}$ for X-output and G2 = 133_{OCT} Y-output. The polynomials of the encoders are often expressed in octal. To traditional polynomial-form these can be easily derived by transforming the number to binary and deriving polynomial coefficients from the binary number. For example number 23 in octal equals 10011 in binary. From this can be derived polynomial $1+z^3+z^4$.

Punctured convolutional coding is used according to the table 2. X and Y refer to the two output of the convolutional encoder. [3]

Table 2: Puncturing pattern and transmitted sequence for the possible code rates [3]

Code rate	Puncturing pattern	Transmitted Sequence
1/2	X: 1 Y: 1	X_1 Y_1
2/3	X: 1 0 Y: 1 1	X_1 Y_1 Y_2
3/4	X: 1 0 1 Y: 1 1 0	X_1 Y_1 Y_2 X_3
5/6	X: 1 0 1 0 1 Y: 1 1 0 1 0	X_1 Y_1 Y_2 X_3 Y_4 X_5
7/8	X: 1 0 0 0 1 0 1 Y: 1 1 1 1 0 1 0	X_1 Y_1 Y_2 Y_3 Y_4 X5 Y_6 X_7

From the figure 4 can be seen convolutional encoder, with code rate 1/2 and polynomials $G_1 = 171_{OCT}$ and G2 = 133_{OCT}. Encoder includes a shift register, which is shifter always, when there's a new bit in the input. The adder is a modulo-2-adder, which gives the remainder of the sum when divided by two. For example with this logic, 1+1=0, 1+0=1 and 0+1=1.

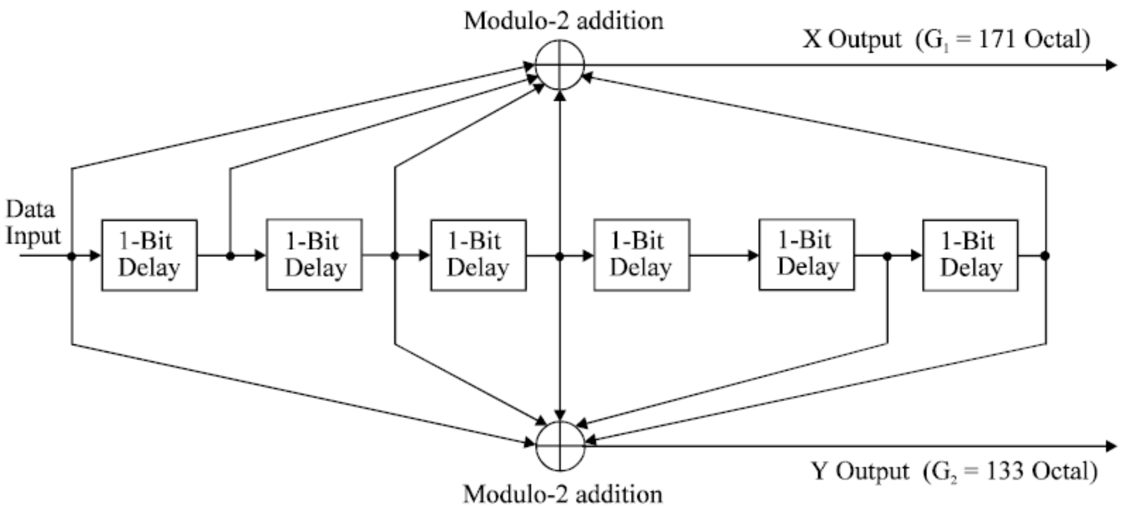


Figure 3-4. The function of the convolutional encoder [3].

To decrease the bit errors, in addition to inner coding, also outer coding is used, which is shortened Reed-Solomon-coding RS (204, 188, t=8), which is derived from the original, systematic RS (255, 239, t=8) code. Each of the 188 byte length packet is RS-coded to achieve 204 byte length RS-coded packet. This allows correction of up to 8 byte errors per packet. [3]

In DVB-T standard, Quasi Error Free-criterion (QEF) is defined, which corresponds to "less than one bit error per hour". This equals in practice errorless data transmission. The threshold value as a bit-error-ratio is 2*10^-4 after Viterbi-inner coding which equals in 10^-11 after Reed Solomon-outer coding. [3]

In addition to inner and outer coding also data interleaving is used. In interleaving the data is rearranged for the transmission. This is how the error sequences and bursts are endured better, because due to the interleaving, the errors are spread more evenly in the data and single errors can be recognised and corrected better than several errors situated close to each other. [3]

All of the sub-carriers are not used to transmission of the data, but part of them are used to pilot signals and transfer parameter signalling (TPS). In 2k-mode, the number of the sub-carriers carrying useful data is 1512 and in 8k-mode it is 6048. Pilot signals are data cells, phase and amplitude of which is known. The receiver uses pilot signals to observe the changes in the channel as a function of time and frequency. The transmission power of pilot signals is greater compared to other sub-carriers. In DVB-T, there are both continuous and scattered pilots. In 2k-mode there are 45 and in 8k-mode there are 177 continuous pilot signals, and they are repeated at the same place in every symbol. The scattered pilots change their place systematically in subsequent symbols with 4 symbol cycles and every 12. sub-carrier is scattered pilot. Scattered pilots place with the continuous pilots to the same carrier every fourth symbol. [3]

Transmission parameter signals transfer the parameters of the sent signal to the receiver. They are not the primary parameter announcers, because the parameters are already known, when the signal is received. TPS-signals are thus used only in special occasions, for example changing parameters or resynchronization. In 2k-mode there are 17 and in 8k-mode there are 68 TPS-carriers and they are situated in every symbol at the same place. Every TPS-carrier located in a certain symbol carry the same TPS-bit. Each OFDM-frame contains 68 OFDM-symbols and, as follows, 68 TPS-bits, which are determined in the following way:

- 1 initialisation bit
- 16 synchronization bits
- 37 information bits
- 14 additional bits for error correction

Out of the 37 information bits 23 are in use at the moment. The rest 14 are reserved for future use and they are set as zero. [3]

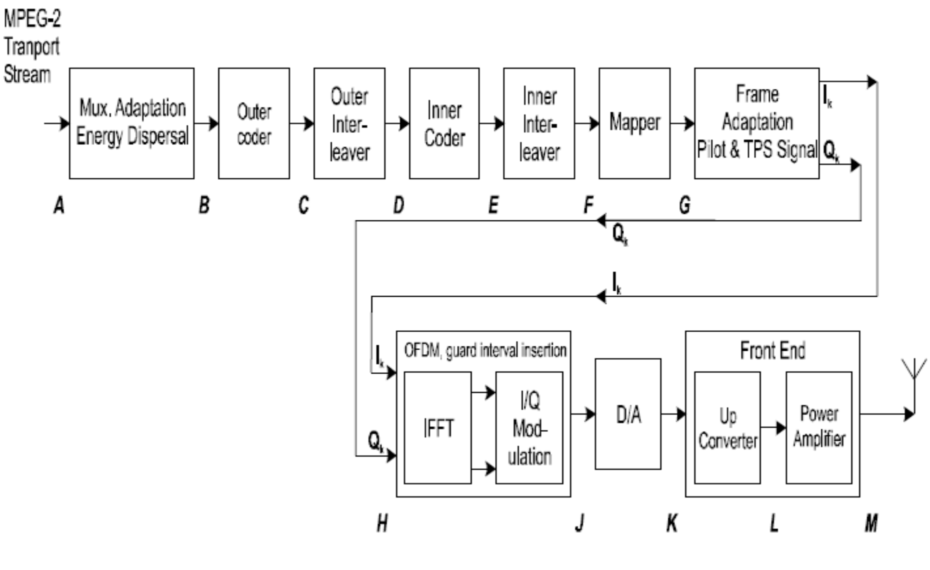


Figure 3-5. Block diagram of the DVB-T-transmitter [5].

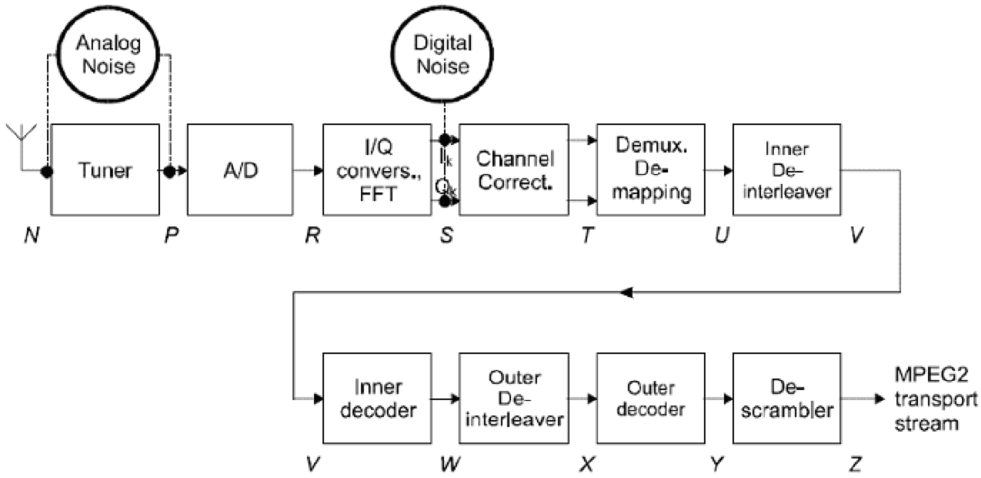


Figure 3-6. Block diagram of the DVB-T-receiver [5].

From figure 5 and 6 can be seen the structure and functions of DVB-transmitter and –receiver. In the following is went through, how the transmission proceeds in the transmitter.

The bit stream to be send is arranged to constant length 188-bit packets, and the order of the packets is randomized to remove correlation (section B). In the outer coding, the data is protected from errors using Reed-Solomon-coding, after which the data is interleaved (sections C and D). After that, the data is transferred to be inner coded, where convolutional coding is used. Then the data is again interleaved (sections E and F).

After this the bit sequence is represented in the baseband as complex symbols, where three different modulation can be used: QPSK, 16QAM or 64QAM (section G). The complex symbols are grouped depending on the used transmission mode to constant length sections, OFDM-symbols. OFDM-frame consists of 68 OFDM-symbols and super frame consists of 4 frame. Pilot- and TPS-signals are added to each of the sections (section H).

Section sequences are modulated with OFDM using 2048 or 8192 (2k- or 8k-mode) sub-carriers. To make receiving easier, guard interval is added to each of the sections (sections J and K).

After this the digital signal is transformed to analog and converted to transmission frequency and amplified (sections K, L and M).

Signal is then sent to the transmission route and in the receiver the signal is converted to usable form with invert operations in inverse order compared to the transmitter.

From the table 3 can be seen the DVB-parameters used in United Kingdom, Germany and Finland.

Table 3: DVB-parameters used in United Kingdom, Germany and Finland [6]

Country	United Kingdom	Germany	Finland
Frequency-band	UHF	UHF&VHF	UHF
Transmission-mode	2k	8k	8k

Guard interval	1/32	1/8	1/8
Code rate	3/4 16QAM & 2/3 64QAM	2/3 & 3/4	2/3
Modulation	suurin osa 16QAM, 2 muxia 64QAM	16QAM	64QAM
MFN/SFN?	MFN	SFN&MFN	MFN
Channel bandwidth	8MHz	8MHZ UHF & 7MHz VHF	8 MHz

3.2 DVB-T2

DVB-T2 (Digital Video Broadcasting – Terrestrial 2) is a second generation terrestrial digital transmission system, which allows more efficient data transmission compared to its predecessor DVB-T. DVB-T2-standard was accepted and published in July 2008. Its primary purpose is to offer transmission route in addition to the Standard Definition (SD) broadcasts, also to the High Definition (HD) broadcasts, which require more capacity. There is added new parameter alternatives as well as new methods to make data transmission more efficient.

The outer coding chosen is BCH-coding (Bose-Chaudhuri-Hocquenghem multiple error correction binary block code) and inner coding used is LDPC-coding (Low Density Parity Check). The code rates used in DVB-T2 are 1/2, 3/5, 2/3, 3/4, 4/5 and 5/6.

In addition to the constellations used in DVB-T, DVB-T2 introduces also 256QAM-constellation. Besides the guard intervals in its predecessors, new usable guard intervals are 1/128, 19/128 and 19/256. Also transmission modes have been added as modes possible are 1k-, 2k-, 4k-, 8k-, 16k- and 32k-modes. The channel bandwidths used in DVB-T2 are 1,7; 5; 6; 7; 8 and 10 MHz. The amount of the used pilot signals is changed as well. In DVB-T scattered pilots consisted 8% and continuous pilots 2,6% of all sub-carriers. In DVB-T2 scattered pilots can be 8%,4%,2% or 1% and continouos pilots 0,35% of the sub-carriers.

New features have been developed for the DVB-T2 too. One of the OFDM’s typical problems is high Peak-to-Average Power Ratio (PAPR), which means, that some of the peak values can be considerably greater than the values typically. These peak values can affect the amplifier by impairing it or even damaging it. They can also result in cutting the signal in high power values, which leads to distortion and disruption to adjacent channels. This problem have been decreased by adding two PAPR-reduction technologies, from which either one or both can be used. In Active Constellation Extension (ACE), the constellation is distorted by moving some constellation points further away to reduce the power of the peak power. The other PAPR-reduction technology is called Tone Reservation, where part of the sub-carriers are reserved to create “anti-peak” where would otherwise develop a high peak. This is how the value of the peak can be attenuated.

DVB-T2 introduces constellation rotation. With this method, the constellation is rotated up to 30 degrees counterclockwise. This adds robustness against multipath propagation in the transmission.

For example, the DVB-T-parameters used in United Kingdom (64QAM, 2k-mode, 2/3 code rate, 1/32 guard interval) and corresponding parameters with DVB-T2 (256QAM, 32k-mode, 3/5 code rate, 1/128 guard interval) allow the data transmission rate to increase from 24,13 Mbit/s to 34,5 Mbit/s when using DVB-T2. [7]

3.3 DVB-H

DVB-H (Digital Video Broadcasting - Handheld) is a digital transmission system, which was developed especially for wireless, handheld reception, like mobile phone. It is closely related to other DVB-standards, especially DVB-T, but it is designed to work with lower electricity consumption, in difficult reception conditions, fast moving environment and its required channel bandwidth is smaller, because the quality of the transmitted audio and video doesn’t have to be as good as for example in the DVB-T standard.

The transmission mode in DVB-H is in addition to 2k- and 8k- mode is also 4k-mode, which provides an additional degree of flexibility for network planning. For 2k- and 4k-modes was added a short, in-depth interleaving, which provides an extra level of protection against short noise impulses.

Time-Slicing technique is an essential part of DVB-H, because it reduces the average power consumption. Each TV-service in DVB-H-signal is sent in bursts, which enables the receiver to be active only while receiving the burst. Rest of the time receiver can be inactive, which could lead to significant power savings on handheld terminals. Time-Slicing is a notable feature in terms of better battery life and warmth balance. DVB-H supports also statistical multiplexing, with which the transmission rate received by every service can be adjusted according to the requirements. [8]

DVB-H-standard didn’t achieve the expected popularity at the market, because almost everywhere the maintenance of the mobile-TV-network based on DVB-H-standard has been quit. In Finland, as almost last country in Europe, mobile-TV-network is still functioning. The main problems of DVB-H are the narrow assortment of the supporting terminals, scarcity of content services and the strong competition of third generation mobile technologies and services they provide. [9]

3.4 DVB-C

DVB-C (Digital Video Broadcasting – Cable) is a digital transmission method, which is used to transmit digital TV-broadcasts in cable network. It was first published and ratified year 1994. The greatest difference between DVB-C and DVB-T is, that DVB-C uses QAM-modulation of only one carrier instead of COFDM. While cable is quite reliable transmission method, DVB-C settles with less error correction than DVB-T. Error correction used in DVB-C is Reed Solomon-coding and no other correction is needed. DVB-C uses also only one interleaving. Pilot-signals and guard interval isn’t needed in DVB-C either. Modulation used can be 16-, 32-, 64-, 128- and 256QAM.

DVB-C2 is a second generation cable-TV-standard perfected from DVB-C. It has added parameter alternatives and spectrum efficiency. It has been changed to use OFDM-modulation because of its benefits. Also error correction and robustness to interference have been developed further by adding new ways of interleaving and effective combination of BCH- and LDPC-coding, but it is also possible to use Reed Solomon-coding. Because of the more effective error correction, DVB-C2 enables even more effective data compression methods and still staying in QEF-area. Modulation in DVB-C2 can be up to 4096QAM. [1]

3.5 DVB-S

DVB-S (Digital Video Broadcasting – Satellite) is a DVB-standard to transmit TV-broadcast through satellite connection. DVB-S was ratified year 1994. It uses Reed Solomon-coding and inner convolutional coding and double interleaving. Modulation used in DVB-S is Quadrature Phase Shift Keying (QPSK).

There is also second generation for satellite digital TV, DVB-S2. It offers three new modulation alternatives, 8PSK, 16APSK and 32APSK. It also has an effective combination of BCH- and LDPC-coding. DVB-S2 has also added adaptive coding and modulation, which means, that coding and modulation can be changed between the frames depending on the state of the transmit channel. [1]

3.6 References

[1] http://www.dvb.org/

[2] http://www.digitv.fi/

[3] http://www.etsi.org/deliver/etsi_en/300700_300799/300744/01.06.01_60/en_300744v010601p.pdf

[4] http://www.conniq.com/WiMAX/fdm-ofdm-ofdma-sofdma-01.htm

[5] http://www.etsi.org/deliver/etsi_tr/101200_101299/101290/01.02.01_60/tr_101290v010201p.pdf

[6] http://www.enensys.com/documents/application_notes/ENENSYS%20Technologies%20-%20DVB-T%20broadcasting%20configurations%20for%20DTTV.pdf

[7] http://www.etsi.org/deliver/etsi_en/302700_302799/302755/01.01.01_60/en_302755v010101p.pdf

[8] http://www.etsi.org/deliver/etsi_en/302300_302399/302304/01.01.01_60/en_302304v010101p.pdf

[9] http://www.mobiilitv.fi

[10] http://www.cisco.com/en/US/prod/collateral/video/ps8806/ps5684/ps2209/prod_white_paper0900aecd805738f5_ps2217_Products_White_Paper.html

[11] http://www.mydarc.de/dk7zb/HB9CV/Details-HB9CV.htm

4 Preliminary Exercises

4.1 Power spectral density of a modulation scheme

In this exercise we investigate the output signal spectrum and adjacent channel leakage for the system described in the figure 4-1.

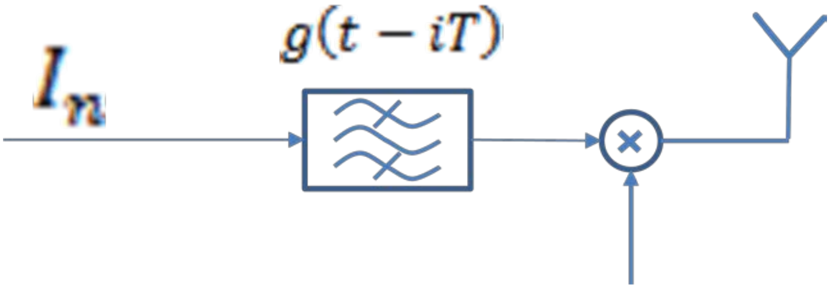


Figure 4-1. Illustration of the modulation process.

a) Derive the power spectral density of the output signal $u(t)$. The $u(t)$ is described as

$$u(t) = \sum_i I_n g(t - iT),$$

where I_n is the information signal with values ± 1 and $g(t - iT)$ is the pulse shaping filter.

Compute the power spectral density of $u(t)$

i. if the pulse shaping filter is a square pulse with length T

$$\mathrm{g(t)} = \begin{cases} 1/\mathrm{T} & |\mathrm{t}| \leq \mathrm{T}/2 \\ 0 & |\mathrm{t}| > T/2 \end{cases}$$

Into the preliminary report include the intermediate steps of the derivation.

ii. if g(t) is *sinc* function from minus infinity to infinity

$$\mathrm{g(t)} = \mathrm{sinc(t\,2/T)}$$

Hint: You can use the duality between the rectangular pulse and the *sinc* function.

In the report it is sufficient if you give the final equation.

b) Investigate the signal spectrum if the signal samples are extended in frequency or in time. In this investigation use the Matlab programs *RectSpectrumComputation.m* and *SincSpectrumComputation.m*.and.

i. Plot the spectrums of the output signal with both shaping filters (rectangular and sinc shaping) on a same figure.

Hint: For plotting that figure you can use default setting in the programs.

ii. Identify the oversampling rate used in computation of the rectangular shaped signal spectrum in *RectSpectrumComputation.m* file.
Change the oversampling to be twice of the current oversampling. Make a plot of the initial power spectrum of the signal and the signal with the new oversampling. How the change of

oversampling has changed the signal spectrum samples? Describe the difference between the two spectrums. Give an "intuitive" explanation on why the spectrums are different? (Hint: In the plot you can select the interval where to show the samples. For instance for the frequency interval 0 … 0.1 MHz you can use Matlab command axis([0 0.1 -80 -60]))

iii. What is the *sinc* filter length in used in the file *SincSpectrumComputation.m*? Change the sinc filter length to be twice of the current filter length. Plot the output signal power spectrum for both of these filter lengths. Describe of what has happened to the signal spectrum. Quantify the differences, for instance how many dB you see the difference,

c) Compute the signal spectrum of a OFDM signal that has 600 data carriers and the OFDM symbol length is 1024 T. The data is allocated symmetrical around the DC but the DC carrier itself is set zero.
   i. Sketch the signal spectrum. What is the distance between the data carriers?
   ii. Use the program *OFDMSpectrumComputations.m* and
      a. for the signal with spectrum exactly 1024 samples wide, plot the time representation of the signal (signal after inverse FFT). In the plot show first 100 samples of the real and imaginary part of the signal (samples from 0 … 100*T).
      b. plot the signal in time if the spectrum is extended from both side with $1024 \cdot (\mathrm{N}-1)/2$ zeros, where N = 8. Plot the real and imaginary part of the signal. Put the plot on the same figure as in the previous case ii.a. In the plot show only signal in the same time interval as in the previous case. (Hint. now the signal is oversampled and for the same interval you have to plot more points).
      c. What is the difference in difference of the time domain signals in case a and b. How the extension of the signal spectrum with zeros impacts the signal presentation in time.
      d. Take the time domain signal from the case b. and extend it with $1024 \cdot (\mathrm{L}-1)$ zeros. (L = 16). Plot the power spectrum of that signal and the signal in the case b above. Show in the figure spectrum magnitude in interval 0…0.1 MHz. (Hint: For better visualization you can use squares and stars for indicating the location of individual frequencies. In the extend case you should have more dense plot of frequencies).

4.2 Adjacent channel interference

By using the signal spectrum plot from previous exercise for all the three modulation cases: rectangular filtered QAM, $sinc$ filtered QAM, OFDM signal. The channel bandwidth is 5MHz and the symbol length $T = 0.2\ \mu s$.

a) Mark on the spectrum plots the co-channel and adjacent channel intervals.
b) By using the program *SignalPowerandAdjacentChannelnterferenceComputation.m* compute the signal in channel power and adjacent channel interference for all these three cases. The adjacent channel interference is computed for the neighboring channel N+1 and for the channel N+2. Collect the results into a table.
c) Compare and comment on how these schemes impact the adjacent channel interference.

4.3 Impact of interference on the receiver

a) Compute the adjacent channel interference if the received signal level is -50 dBm, the co-channel interference level is -73 dBm and the observed SIR = 20 dB.
b) Compute the adjacent transmitter power if ACLR = -40 dB, ACS = -25 dB and the transmitter is located d=100 from the receiver. Attenuation is modeled as power law path loss y $d^{-3.5}$. The received adjacent channel interference level is computed in the case a).

4.4 SINR requirements different for different modulation schemes.

Find the bitrates with Finnish, United Kingdoms (16QAM) and Germanys DVB-T parameters from the standards.

5 Lab Measurements

5.1 *Field strength measurement*

Measure the received TV signal power in channel 32 (562 MHz Espoo transmitter). You can measure the signal strength by the spectrum analyzer Tektronix RSA 6114A. Details of how to use measurement equipments are given in the appendix.

a) Measure the received signal strength from the rooftop antenna.
b) Received signal from the indoor antenna
   a. For inside antenna make the measurements by directing antenna in different directions in horizontal plane.
   b. For direction where the signal is strongest make measurements by changing polarization of the antenna (rotating antenna in vertical plane).
c) Measure the impact of the measurement setup on the signal strength.
   a. Measurement set up is described in Figure 5-1 right. How much is the signal strength reduced compared to the rooftop received signal.
   b.

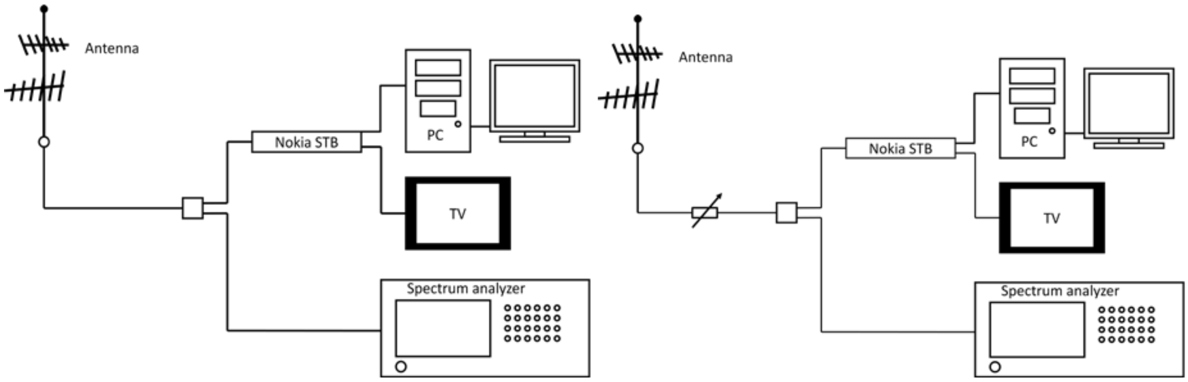


Figure 5-1. Connections for measuring the signal strength from the antenna, with and without attenuator..

5.1.1 Post laboratory exercises

a) What is the difference between the signal received from the rooftop antenna and from the inside antenna.
b) Plot the signal strength as a function of antenna direction.
   a. Comment on the figure. How much the signal strength is changing? From which direction the signal is strongest?
c) Plot the signal change in after rotating the antenna in vertical direction.
   a. What is the impact of the antenna polarizations? How much the impact?
   b. Find from the literature what is the polarization of the TV signal from the Espoo TV transmitter? Does this information comply with your measurements?

5.2 *Connection quality measurements.*

a) Measure the BER of the signal from rooftop antenna by changing the attenuation in the received chain (see connection in Figure 1 right). You can measure the signal by the Nokia Mediamaster 9730C set-top box and HyperTerminal program in the PC. Details of how to use measurement equipments are given in the appendix. Observe and comment on the of the television image visually in different attenuation levels.

b) Measure the BER and MER with two different modulations 16 QAM and 64 QAM. Generate the TV signal by the signal generator (you can use Rohde&Schwarz SMBV100A). Use the signal settings

Table 1 Parameters for TV signal in BER measurements

Variable	Value
Mode	8k
Guard interval	1/8
Channel BW	8 MHz
Code rate	2/3

In the measurements you can use Tektronix RSA 6114A with DBA-program. DBA program can be found in the Windows workspace of the spectrum analyzer.

a) Take up the BER values after the inner Viterbi decoder and after outer RS decoder.
b) Draw or take screenshots also the constellation diagram given by the spectrum analyzer from few different stages.

5.2.1 Post laboratory exercises

a) For both modulations plot the BER a the decoders output
b) Comment on the difference of the curves. What differences you observe? Explain the reasons for the differences?
c) The TV Tx power is 50 W and height 300 m and distance to the measurement place is 10800 m.
    a. Use Okomura-Hata channel model and compute the expected signal level at the rooftop antenna.

$$a = 69.55 + 26.16lg\left(\frac{f}{MHz}\right) - 13.82lg\left(\frac{h_{eff}}{m}\right) - \frac{c(h_r)}{dB} + \left(44.9 - 6.55lg\left(\frac{h_{eff}}{m}\right)\right) lg\left(\frac{d}{km}\right)$$

f is the frequency in MHz, $\mathrm{h_{eff}}$ is efficient antenna hight, d is the distance in km.

$c(h_r) = 3.3\left(\lg\left(11.75\frac{h_r}{m}\right)\right)^2 - 4.97$ is the term characterizing path loss in the urban environment.

    b. From the measurements you know what is the signal level where the connection is lost, compute the link budget for that level. How big would this cell be (use Okomura-Hata model from above).

5.3 Interference measurements

a) Measure the impact of the co-channel by changing the co-channel signal level.
    a. Make the connection as described on the figure 5-2.
    b. Set the TV signal level at the first signal generator to be -70 dBm.
        i. Measure the received TV signal level at the spectrum analyzer.
        ii. Set the co-channel interfering signal level to be -90 dBm.
            1. Measure the received co-channel signal level at the spectrum analyzer.

c. Connect both the signals and start to increase the co-channel signal level.
    i. Measure the BER of the TV signal as the function of the interference level.

b) Adjacent channel interference level measurement.
    a. Use the TV signal as in the case the case 3.1. TV signal level -70 dBm.
    b. Set the interfering signal to transmit on the first adjacent channel.

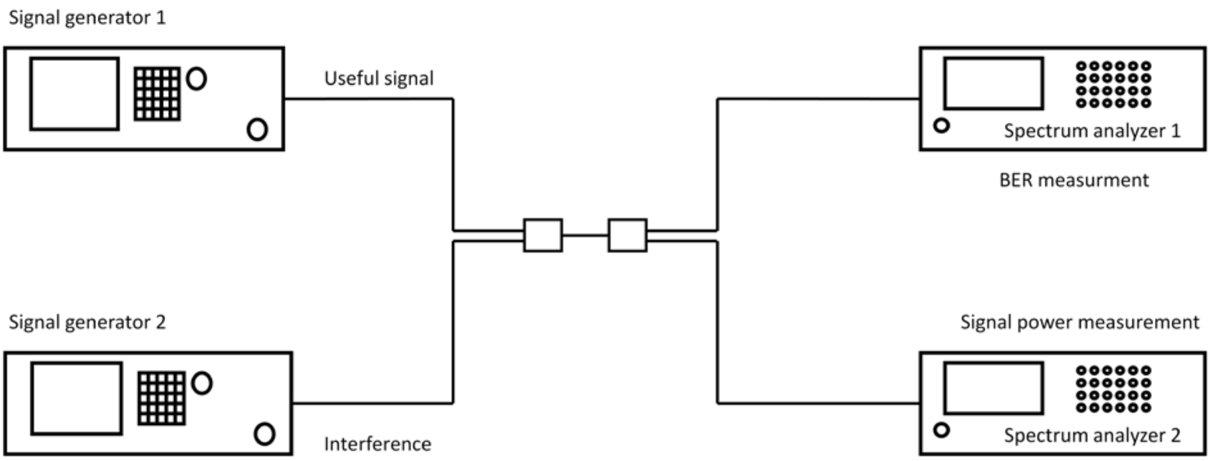


Figure 5-2. Set up for interference measurements.

    i. Start to increase the adjacent channel interference level and measure the resulting BER (with spectrum analyzer 1) and the adjacent channel power (with spectrum analyzer 2).
    c. Repeat the adjacent channel measurement by setting the adjacent channel transmitter to use fifth adjacent channel.

5.3.1 Post laboratory exercises

a) Plot the BER as a function of the interference level for both co-channel and adjacent channel interference.
b) At what level the co-channel interference forces the BER to exceed 1e-3.
c) What is the adjacent channel interference level when BER is 1e-3.