10

DSSTV transmission systems

10.1 Redundant Data File Transfer

This communication mode uses *phase shift keying (PSK)* modulation. The simplest PSK modulation changes subcarrier between two phase states *(BPSK – biphase shift keying)* and these states corresponds to level of logical zero or one. This is used for example for teletype mode PSK31.

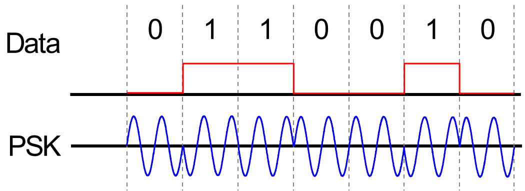


Figure 10.1: Two-phased digital PSK modulation.

RDFT uses similar principle but much more extended. The signal is composed from eight subcarriers from 590 Hz to 2200 Hz with 230Hz steps. Each subcarrier uses nine modulation states – eight informational states and one state with *no change* meaning. The data from inner encoder are used for phase assignment.

The first step in modulation process is to take a cosine of modulation angle plus 1400Hz subcarrier angle. In next steps the energy around 1400 Hz is isolated and translated onto right subcarrier. Then the subcarriers are compiled together and resulting spectrum of signal is in **fig. 10.3**.

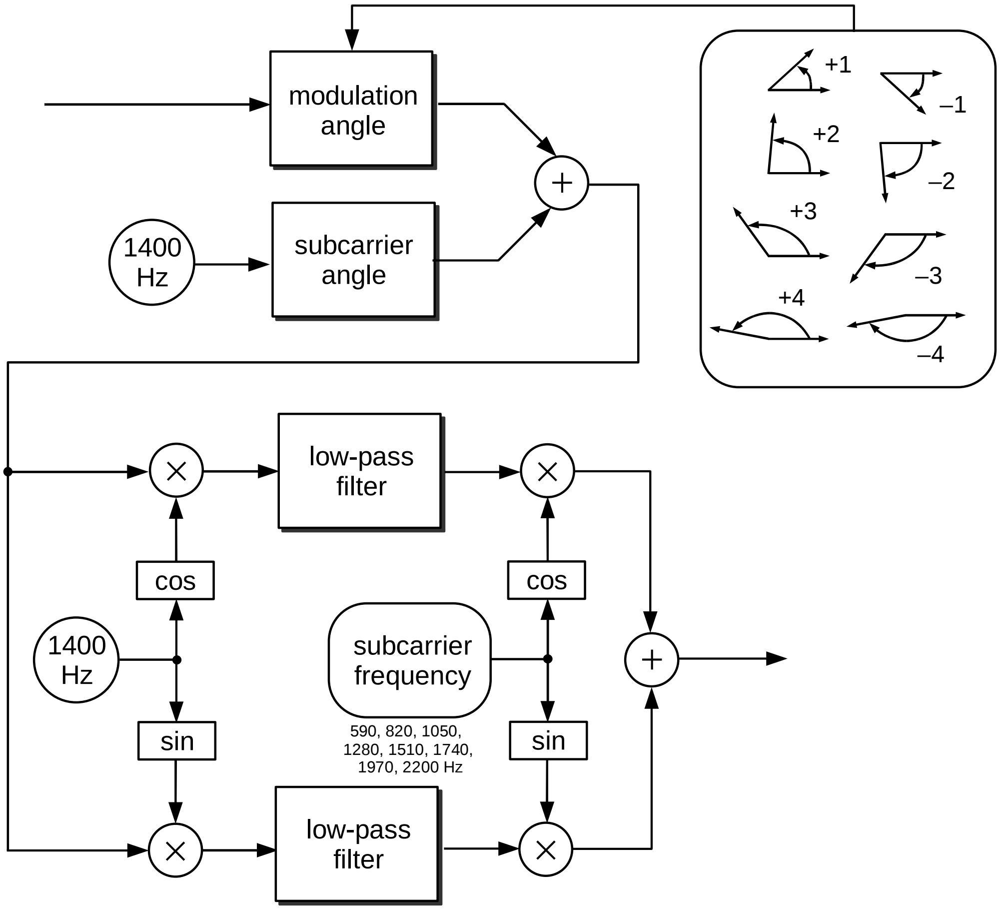


Figure 10.2: Block diagram of RDFT modulator for one subcarrier.

The signal contains two levels of error-coding. The outer coding scheme use RS code $(306, x)$, where x is set by level of error control, see **tab. 10.1**. The symbol numbers, produced by the outer encoder, are the input to the inner encoder. The inner coding scheme uses RS((8, 4), where is 50% redundancy. All 8 symbols are used for phase settings for each of 8 subcarriers, so the inner code-block is transferred paralely. The decoder of inner code on reception side is able to correct whole block if 6 of 8 symbols are transferred without error.

Operator can choose one of four modes, in all cases the modulation speed is same 122,5 Bd, but the level of error control differs.

Table 10.1 are parameters of RDFT modes. You can choose lower error control level when band conditions are good or higher level in case of bad conditions and big interference. The redundant data consume 70 % of all transferred data for *Wyman 14*, so there is possible to apply an extensive error-correction. The Wyman 13 is recommended for long-distance contacts and Wyman 12 for intracontinental QSOs.

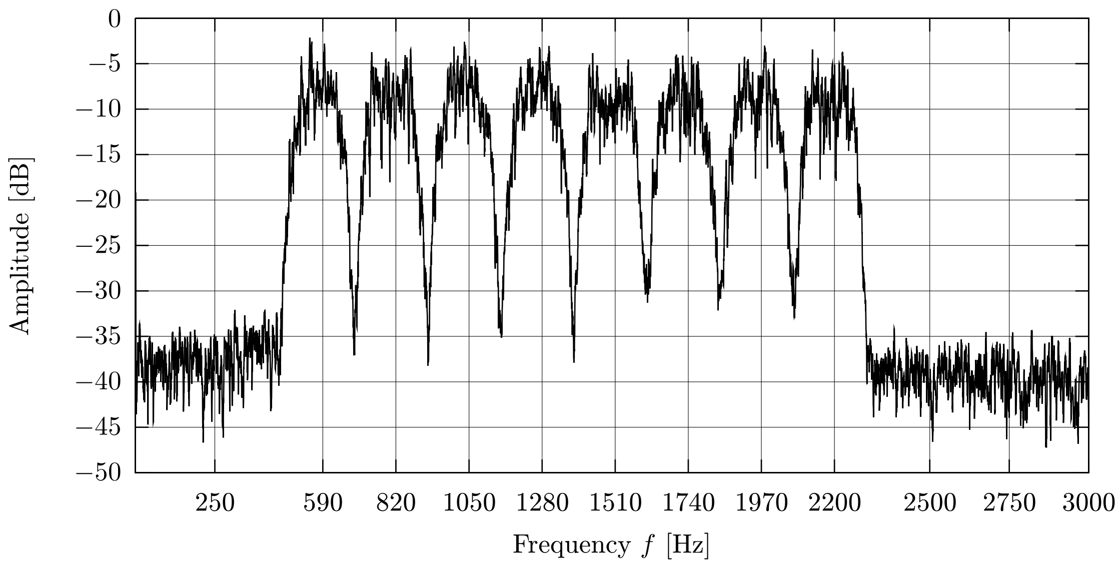


Figure 10.3: RDFT frequency spectrum

RDFT mode	Subcarrier number	RS code	Redun-dancy	Maximum errors	Band-width	Bit rate	Symbol rate
Wyman 11	8	(306, 274)	10 %	16	1840 Hz	866 bps	122,5 Bd
Wyman 12	8	(306, 242)	20 %	32	1840 Hz	765 bps	122,5 Bd
Wyman 13	8	(306, 178)	40 %	64	1840 Hz	563 bps	122,5 Bd
Wyman 14	8	(306, 92)	70 %	107	1840 Hz	291 bps	122,5 Bd

Table 10.1: RDFT communication modes.

Transferred data block consists of three parts:

▷ The first is *LEADER*, it uses always same modulation scheme and error-coding. It contains RDFT mode identification and it is used for detection of two parameter. The first parameter is a tuning deviation in Hz, because most SSB transceivers have smallest tunning step 10 Hz it is not possible to tune accurately. The next parameter is clock rate difference, it is caused by inexact sample rates of sound cards and there is also small difference on receiver and transmitter side. Both these parameters are dynamically found during transmission and they are used in demodulation process.

▷ The next part is *CODEBLOCK*, it is sequence of data frames of transferred file with redundancy symbols for error correction.

▷ The transfer ends with *TRAILER*, it contains mode identification like the first part.

The average bit rate of transfer is about 736 bps (92 bytes per second).

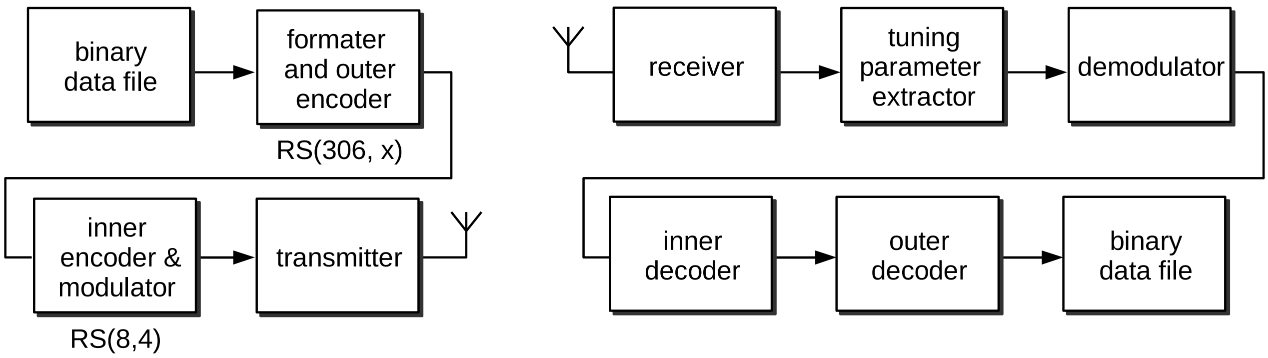


Figure 10.4: RDFT communication channel.

The input of demodulator are samples of RDFT signal, the output are some phase states of each subcarrier. The block circuit of demodulator is in **fig. 10.2**. The *delay* block provides a delay of one symbol period. Subtracting the angle values separated in time by one symbol period is the "differential" portion of this "differential phase demodulator". The *average* block averages 24 adjacent differences and divides by the unit phase step, to produce the final demodulator output. This averaging helps reduce the intersymbol interference produced by the low pass filters.

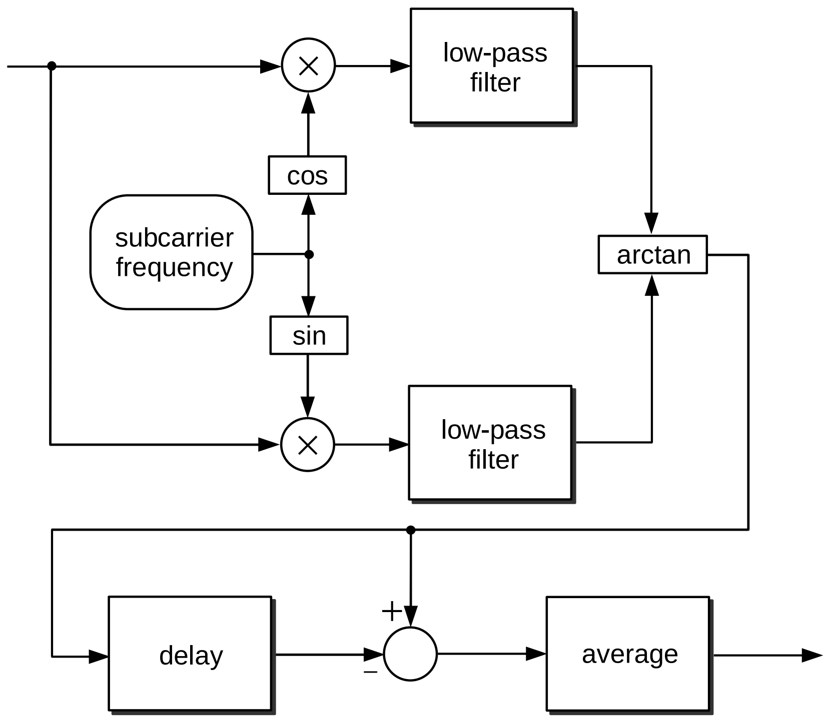
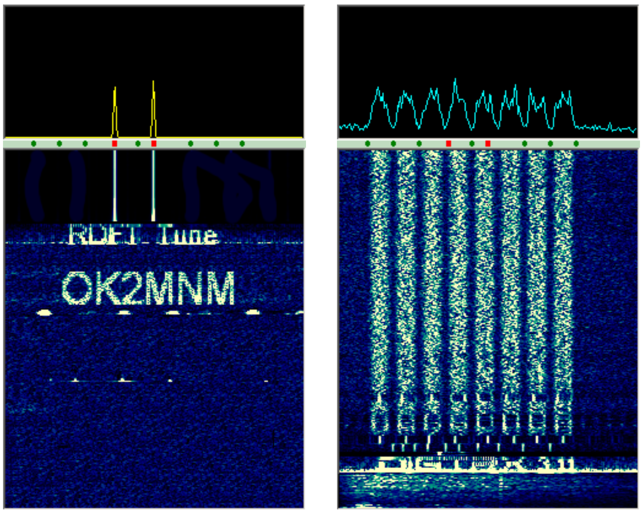


Figure 10.5: The RDFT demodulator block diagram.

10.1.1 RDFT operations

The audio level of sound card is the most important setting on TX side. It must be inside linear range, it is usually the middle half of its dynamic range. On RX side should be sound card level adjusted too. The input signal should not be overdriven, it causes unwanted nonlinearities.

The stations should be very precisely tuned to each other for making successful contact. The station before the data transfer should send tunning signal. This signal consist of two tones 1180 and 1520 Hz. These frequencies are labeled on spectroscope, so you can fine tune and align the frequency peeks with the marks.



Tunning signal and station id. Spectroscope with data transmission.

Figure 10.6: The tuning spectrospoce in DIGTRX.

Following operations must be done for transmission and reception of images:

1. Original data file is processes, in case of images the resolution and compression level are set. Then is generated WAV audio file, which contains a audio signal for radio transmission. The time spent to signal generation is derived from input file size and computer configuration. It can took form a second or two (2GHz and faster CPUs) up to several minutes on slower systems (400 MHz).
2. The audio file is played and transmitted. The reception station records sound and store it on a hard drive.
3. The software process recorded WAV file and reconstructs original file. This step is also computationally very demanding and the time needed for decoding depends not only on the volume of data and processor speed, but also on how

much is necessary to use the error-correction algorithm. This step may take several minutes on a slow computer, on a 2GHz machine it takes 15 seconds.

Barry Sanderson, KB4VAK, developed programs for RDFT encoding. These program are command line driven and it is available as open source under GNU GPL license. So programmers can implement it to several computer platforms. So thanks to open source idea, there is few programs where is the RDFT mode available.

RDFT presentation in Dayton conference: http://replay.web.archive.org/20080528090630/http://www.svs.net/wyman/examples/hdsstv/

10.2 HamDRM system

Communication system *HamDRM* is derived from open standard Digital Radio Mondiale [[error 2]], which was created for digitalization of radio broadcast on medium-wave and short-wave bands. Normal DRM use bandwidth 4.5 kHz to 20 kHz for sound quality similar to FM broadcast on VHF. The hamradio version *HamDRM* was created by Francesco Lanza, HB9TLK. It is modified for usage in SSB channel with 2.5 kHz bandwidth. HamDRM can be used for image and data file transfer and also for voice communication, so it should be competitor for analog SSB in future.

The used modulation is *COFDM (Coded Orthogonal Frequency Division Multiplex)*, which has maximal utilization of communication channel. The Reed-Solomon code is used for error correction.

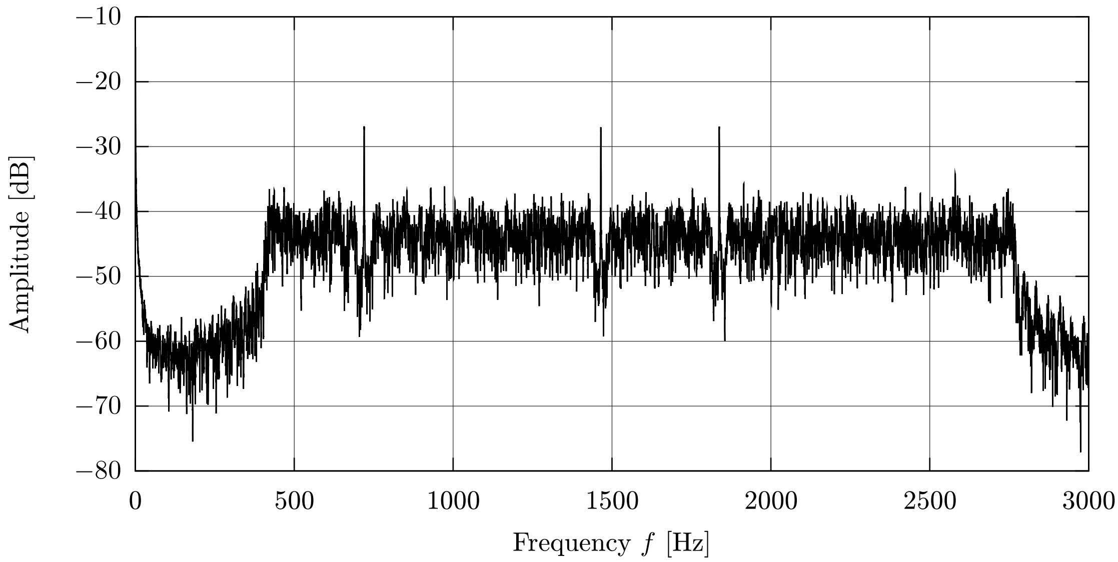


Figure 10.7: The frequency spectrum of HamDRM system.

The OFDM signal consists of a huge number of subcarriers in baseband. There are from 29 to 57 subcarriers in case of HamDRM. An each subcarrier is modulated independently with quadrature amplitude modulation (QAM) and together with error-correction code creates COFDM. This modulation is well resistant to phase distortion, attenuation, selective fading and pulse interference. The used modulation techniques are described later in **section 10.2.2**.

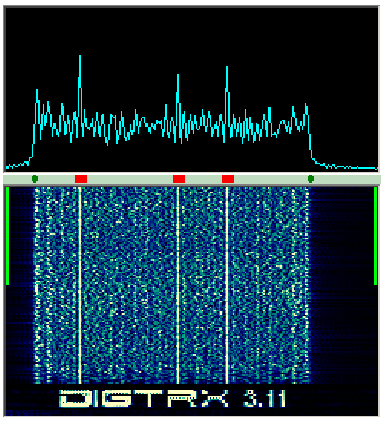


Figure 10.8: The tuning spectroscope in DIGTRX. Note three frequency peeks, that is used as guidelines for proper tuning.

Unlike RDFT, which needs 3 partial operations – coding, recording of broadcast and decoding, the HamDRM doesn't transmit data in whole block, but the file is divided into separate segments, so the image can be decoded and displayed during transmission.

HamDRM can be used in three basic modes. The *Mode A* allows the fastest transmission, but does not protect against the negative effects caused by selective fading. The *Mode B* is slower than the first mode, but is resistant against the negative impacts and it's much more robust. The last mode is *Mode E*, which is designed for communication through a channel with large delay and Doppler effect.

The QAM modulation is used with 4, 16 or 64 states. Modulation QAM-64 is the fastest but it needs a very good level of signal-noise ratio, at least 18 dB. Modulation QAM-4 is slower, but is more resistant to interference and requires a lower signal-noise ratio, about 5–6 dB. Minimum SNR for QAM-16 should be about 8–10 dB. The selection of modulation depends on an user and an actual conditions prevailing on the band. Other HamDRM features that can be set by user are following:

▷ *Interleaving* is used for change of symbol sequence, it is a way to arrange data in a non-contiguous way to increase performance. The long interleave has 2 seconds, it supports better error-correction but causes longer delay during decoding. The short interleave take 400 ms.

- ▷ *Bandwidth* can be changed to 2.3 kHz or 2.5 kHz. A narrower SSB filter can be used for lower bandwidth, but transfer speed is little lower.
- ▷ *Amount of instances* is value that gives number of file repetitions during transmission. If there is more than one instance then all segments will be repeated and the error parts can be corrected automatically on reception side during second or third instance. The number of instances makes transmission time longer.
- ▷ *Leadin* is broadcast at beginning of transmission. This initialization is used to receiver synchronization, extra time allows better synchronization and automatic set up of reception settings.

Details of the mode and its parameters, along with the call sign is broadcast throughout the transmission with QAM-4 modulation, so it is possible to tune to signal during transmission, but the complete data will be received if at least one complete instance of the transferred file is received.

Mode	**Bandwidth**	**Number of subcarriers**	**Level of MSC FEC**	**Transmission speed [bps]**		
				QAM-4	**QAM-16**	**QAM-64**
A	2,3 kHz	53	normal	1480	2710	4170
			low	1900	3470	5340
	2,5 kHz	57	normal	1760	3220	4960
			low	2260	4130	6360
B	2,3 kHz	45	normal	1070	1950	3000
			low	1370	2500	3850
	2,5 kHz	51	normal	1270	2320	3570
			low	1630	2970	4580
E	2,3 kHz	29	normal	690	1270	1950
			low	890	1620	2500
	2,5 kHz	31	normal	820	1510	2320
			low	1060	1930	2970

Table 10.2: The parameters of HamDRM modes and their transmission speed.

Parameter selection of HamDRM modes affects the transmission performance, and hence the transmission speed, which depends on the settings, see **table 10.2**. The two corresponding stations should not communicate with each other in the same mode. E.g. station X has a considerable local interference, so station Y sends in a more resistant mode, but Y hasn't this problem, so X can easily transmit in a faster but less resistant mode.

If the transfer of some segments fails completely, it is not lost at all, because your QSO partner can send *bad segment report (BSR)* and you can resend bad segments again. It's important to send BSR in same mode. The repeated segments can be received third station too and if have not all segments received it can complete whole data. When band conditions are really bad and part of resend fails again, it is possible to generate the new BSR, so the amount of transfered data will be lower in next resent.

The DRM transfer consist of three channels – MSC, SDC and FAC. Each channel is dedicated for transmission of certain data or service information and also for each is used different coding and modulation scheme.

MSC – *Main Service Channel* contains data for all services of DRM multiplex. The multiplex can contain one to four services and each can transfer data or service information.

FAC – *Fast Access Channel*, is support channel. It uses QAM-4 and broadcast callsign, DRM mode identification (band spectrum occupancy, interleaving, mode of MSC and SDC modulation,...).
FAC channel with service information transfer packet with 40 bit size:

- ▷ 2 bit FRAME-ID, identifies a frame in a superframe, value 0, 1, 2
- ▷ 1 bit Spectrum Occupancy (2.3 / 2.5 kHz)
- ▷ 1 bit Interleaver Depth (400 ms / 2 s)
- ▷ 1 bit MSC Mode (QAM-16 / QAM-64)
- ▷ 1 bit Protection Level (amount of FEC used)
- ▷ 1 bit Audio/Data
- ▷ if *audio* is used, then follows:
    - ▷ 2 bit, audio codec: LPC, unused, SPEEX;
    - ▷ 1 bit, text flag;
- ▷ if *data* is used, then follows:
    - ▷ 2 bity, Packet ID;
    - ▷ 1 bit, extended MSC mode (QAM-4);
- ▷ 21 bits, Label, consisting of 3×7 bit ASCII characters (9 characters in superframe)
- ▷ 1 bit, dummy
- ▷ 8 bitů, CRC, used polynomial $G(x) = x^8 + x^4 + x^3 + x^2 + 1$.

SDC – *Service Description Channel* contains information of MSC decoding scheme and broadcast service attributes during multiplexing.

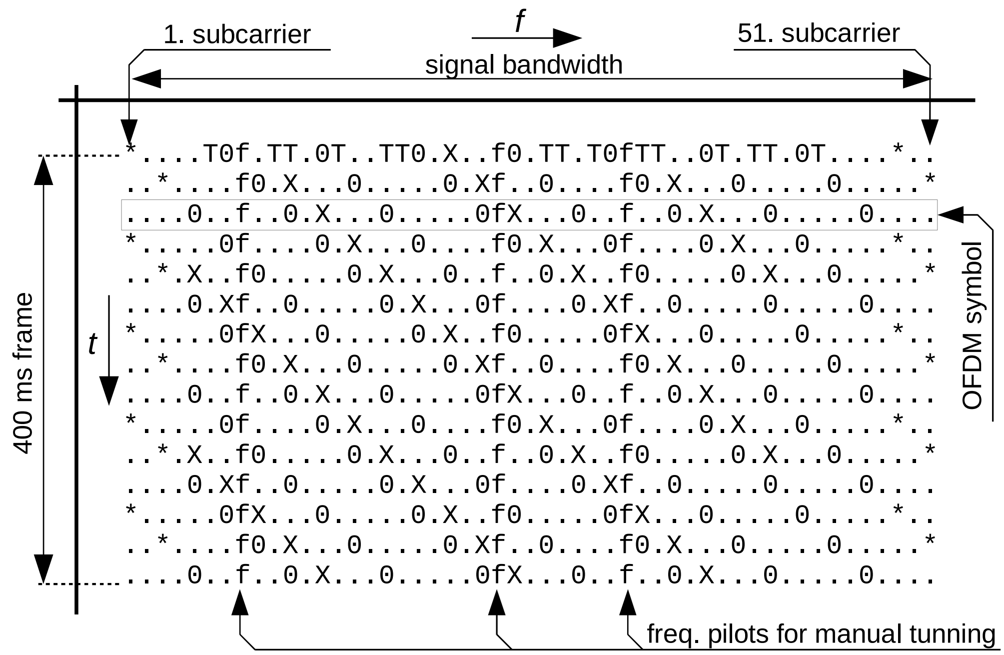


Figure 10.9: Example of frame for *Mode B*, spectrum occupancy 2.5 kHz. Legend: `:` – DC-carrier, `.` – MCS cells, `X` – FAC cells, `T` – time pilots, `f` – frequency pilots, `0` – scattered pilots, `*` – boosted scattered pilots.

10.2.1 Comparison of HamDRM and RDFT

There are several software products for RDFT and HamDRM, but preference of users inclines to HamDRM. Main reason for HamDRM popularity over RDFT are:

- ▷ it is possible to decode and display image during transmission;
- ▷ transfer speed is better up to 3×;
- ▷ HamDRM continuously broadcasts station identification, so receiving operator can start reception and direct yagi;
- ▷ thanks to several instances, it isn't necessary to record the transmission from beginning to end;
- ▷ when reception failed, only bad segments can be repeated, not whole transmission;
- ▷ main disadvantage of HamDRM is, that the powerful PC configuration and OS better then Windows 2000 is a must.

10.2.2 Quadrature amplitude modulation — QAM

Quadrature amplitude modulation (QAM) uses amplitude and phase modulation together. HamDRM for each subcarrier (OFDM cells) can use several modulation schemes, which differ in number of modulation states – QAM-4, QAM-16 and QAM-64.

The number of modulation states QAM-m is divided into $\sqrt{m}$ states for phase keying and $\sqrt{m}$ amplitude levels. Thanks to multistate modulation it is not required so huge bandwidth, on the other hand, a growing number of states of modulation makes the signal less resistant to interference.

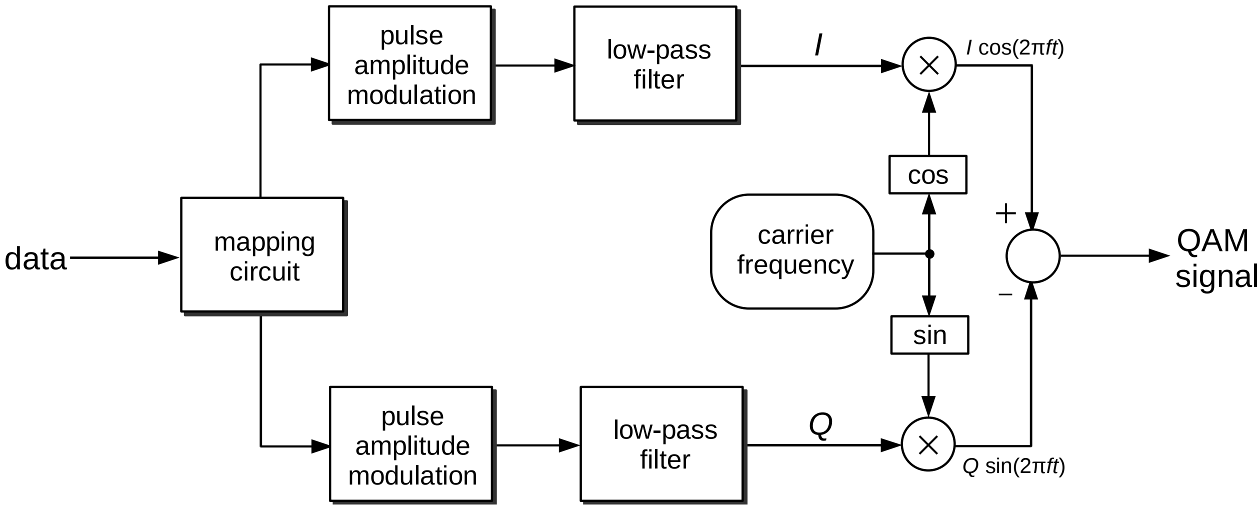


Figure 10.10: The QAM modulator.

An modulation state is created from combination of amplitude and phase, which can define a bit word of length l. For QAM-4 is the word length $l = \log_2 m = \log_2 4 = 2$, for QAM-16 is $l = 4$ and for QAM-64 it is 6. The modulation changes between these states:

$$A_k = 2k - 1 - \sqrt{m} \quad \text{pro} \quad k = 1, 2, ...\sqrt{m}.$$

E.g. for QAM-16 levels are $-3, -1, 1, 3$.

The signal, which can be presented like

$$S_k(t) = A_k \cos(2\pi f t + \phi_k)$$

has 16 combinations of amplitudes A_k and phases ϕ_k.

The block diagram of QAM modulator see in **fig. 10.10**. Now, we describe how QAM-16 modulates data sequence $N = \{0,\ 13,\ 5,\ 2,\ 10,\ 7,\ 6,\ 5,\ 1,\ 15\}$. The result is **fig. 10.12**. The information words with 4bit length are divided on two parts in mapping circuit and first 2bit combination is coded in *pulse amplitude modulation (PAM)* into one of four levels. The way how to code input bit quaternion $\{i_0, i_1, q_0, q_1\}$ is defined by *constellation diagram*, **fig. 10.11**. E.g. for input 0 it is $i_0 i_1 = 00$, $q_0 q_1 = 00$ and this corresponds to $I = 3, Q = 3$, the next value

13, in binary 1101 corresponds $i_0 i_1 = 11$ output $I = -3$ and for $q_0 q_1 = 01$ output $Q = -1$, etc.

The results of PAM are pulses with given amplitudes and they are filtered with low-pass filter for the bandwidth reduction and for in phase path I and similarly for quadrature path Q. The I and Q are input signals for modulators with carrier frequency f. This way there is a phase of 90° between them. Output signal is made by joining of both paths together:

$$S_k(t) = I_k \cos(2\pi f t) - Q_k \sin(2\pi f t).$$

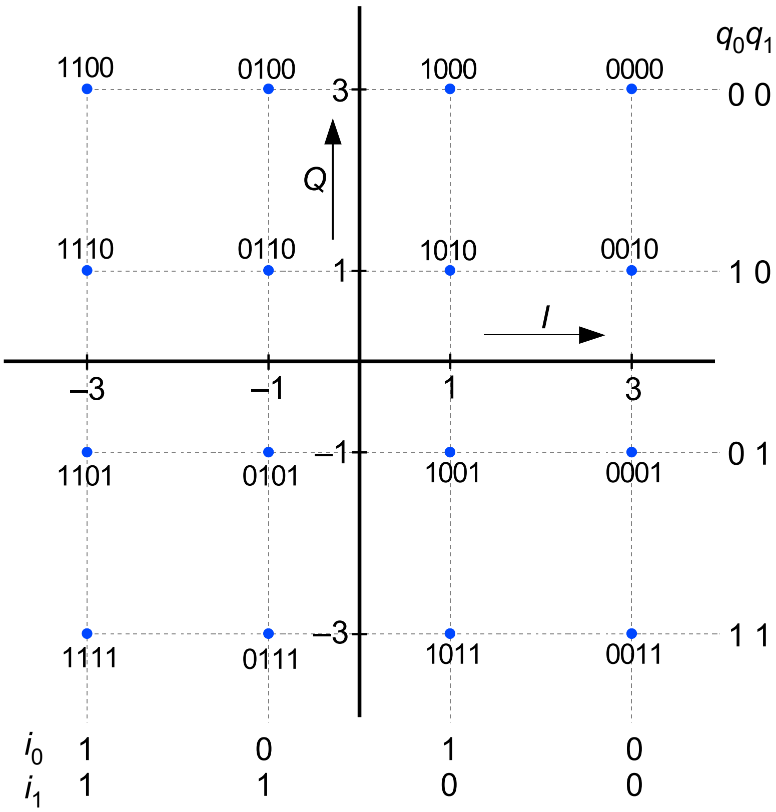


Figure 10.11: The constellation diagram for QAM-16 with bit order $\{i_0, i_1, q_0, q_1\}$ used in DRM.

10.2.3 Orthogonal frequency-division multiplexing — OFDM

OFDM is a representative of the modulation scheme with multiple carriers *MCM (Multicarrier Modulation)*. Thanks to its properties the OFDM found application in many modern technologies, i.e. ADSL, WiFi (IEEE 802.11a/g) networks, WiMAX and standards for digital broadcast and terrestrial digital television DVB-T, etc.

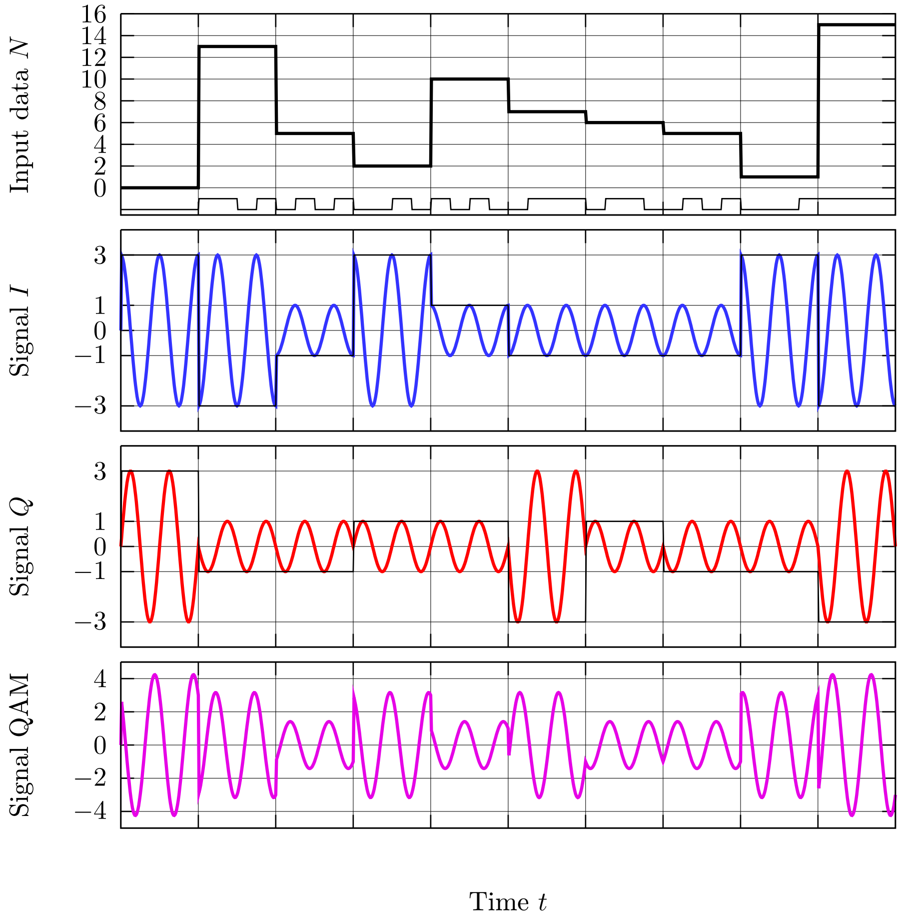


Figure 10.12: The example of QAM-16 modulation for input data sequence.

OFDM has very good spectral performance and it is resistant to pulse interference, because transfered information is dispersed in wide frequency spectrum, so interference disturb only few nearby symbols. It's also resistant to inter-symbol interference, fade outs caused by multipath spreading and has low sensitivity to errors in time synchronization.

The OFDM generates a huge number of subcarrier waves and in case of HamDRM there are for best performance *only* 57 subcarriers. Many other applications like digital video broadcast or wideband data communication uses hundreds or thousands of subcarriers! These subcarriers have very small distances, even those, that the overlap the range of others. An example of OFDM spectrum is in **fig. 10.13**, as spectrum of each subcarrier is considered the spectrum of rectangular signal, which is expressed by $\sin(x)/x$ function.

The subcarriers has exact distances, so maximal level of spectrum of each subcarrier is null in maximal levels of other subcarriers, so they are mutually *orthogonal*.

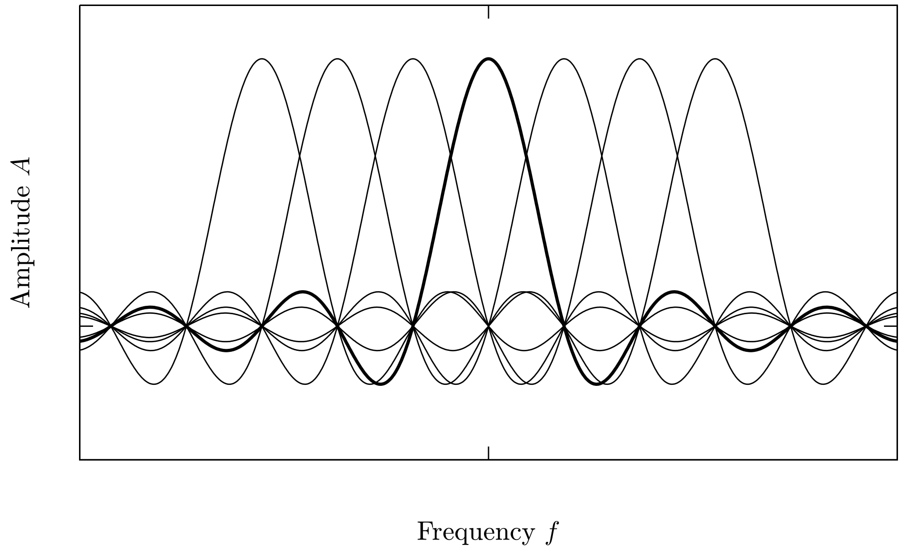


Figure 10.13: The frequency spectrum of orthogonal carrier waves.

10.2.3.1 OFDM transfer

The modulator block diagram is in **fig. 10.14**. Input data stream comes to serial-parallel converter and it is cyclically distributed to a larger number of parallel components. The parallel component transmitted simultaneously creates a complete OFDM symbol. Components are also modulated to the orthogonal system of N subcarriers, the frequencies are distributed to ensure their orthogonality. Subcarrier waves in our case use modulation QAM-4, QAM-16 or QAM-64, but for some other applications there are used multiphase BPSK or QPSK.

A signal processor provides modulation of huge number of subcarriers, in our case it is software, which implements algorithms for inverse discrete Fourier transform (DFT^{-1}). Because DFT algorithm has big computing complexity there is used its faster variant *FFT (Fast Fourier Transform)*. The inverse FFT (FFT^{-1}) transform input data from frequency domain to time domain. The process on a receiver side vice-versa use direct FFT to obtain individual subcarriers.

Two data stream are outputs of FFT^{-1}, which are converted with digital/analog converters on two analog signals. Then these signals are modulated to main carrier and there is a phase of 90° between them. The Re signal presents amplitude component and Im signal phase component. Both joined together creates transmitted OFDM signal.

Everything on reception side goes in opposite way. The received signal is amplified and converted to lower frequency. Then signals Re and Im go through low-pass filters to analog/digital converters and data from them is processed by DSP with

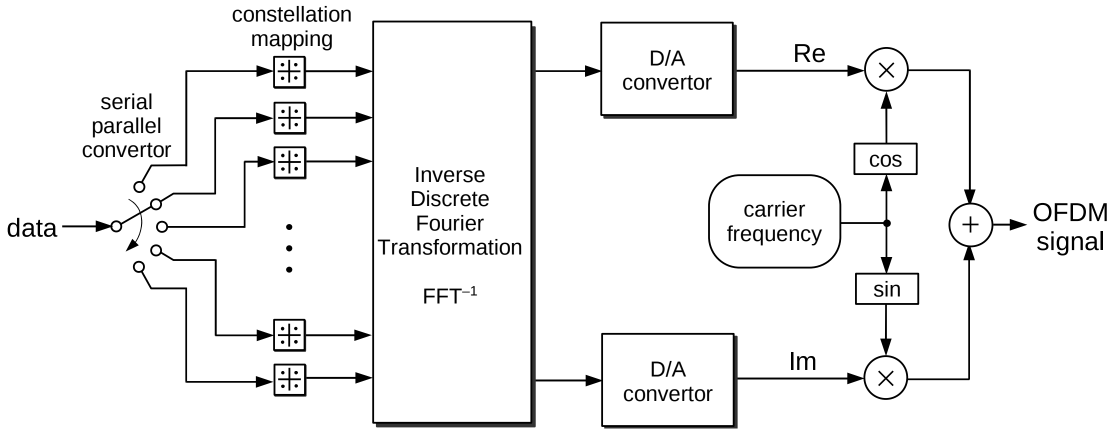


Figure 10.14: The OFDM modulator use fast Fourier transform (FFT^{-1}) for making a huge number of modulated subcarriers.

direct FFT and divided into individual subcarriers. The output data are compiled in parallel-serial converter.

10.3 DSSTV software selection

There is several programs available supporting HamDRM and RDFT.

Software	RDFT	HamDRM	Web page
DIGTRX 3.11	⋆	⋆	http://www.qslnet.de/member/py4zbz /hdsstv/HamDRM.htm
DigiACE V1.9	⋆		http://homepage.ntlworld.com/mhemmerson/
DigPAL		⋆	http://www.home.bellsouth.net/p/PWP-hampal
EasyPAL		⋆	http://vk4aes.com/
HamPAL		⋆	http://www.home.bellsouth.net/p/PWP-hampal
RDFT	⋆		http://www.svs.net/wyman/examples/hdsstv/
RXAMADRM (Linux)		⋆	http://pa0mbo.nl/ties/public_html /hamradio/rxamadrm/index.html
SSTV-PAL Multimode	⋆		http://f6baz.free.fr/FTP/SSTVPalPlus/
WinDRM		⋆	http://n1su.com/windrm/

10.4 Making QSO

Digital SSTV is not spread too far. There are few station found sporadically on the 14MHz band. But there is also working party of German stations on the 3.7MHz band around the frequency 3.733 kHz almost daily in the evening. Stations use only HamDRM system. Listening to their signals is a good opportunity to try DSSTV reception and get some practice with it, also try to make contact. After that, you already know how there is used special modulation schematic and error-correcting coding it is important to see if it at all works and how. Will be there SSTV digitalization boom?

Some opponents of digital video broadcast claim, that in conditions where we can receive noisy, but still usable analog TV signal, the digital TV cannot be received at all. And same argument can say opponents of digital-SSTV. When there are good conditions, it is possible only to tune on channel, images are received automatically and operator should not do anything. When interference gets stronger and signal weaker there can help more data instances or bad segment report and additional repetition of bad segments. But when we only guess HamDRM signal drowned in high noisy level the reception is impossible.

The DSSTV traffic can be found on band near the centre of SSTV activity. Also hamspirit rules should be observed and we should be considerate to another traffic on the band. Sometimes it takes a little tact to explain to uninformed station, that the strange rattling sound is the digital signal from your QSO partner.

A CQ call can be done by sending picture on free frequency. HamDRM during transmission broadcast station id, so is you don't receive complete data you can see what station is transmitting. After the end of transmission you can call the station by voice.

For reception confirmation or short message transfer there is used waterfall messages – messages displayed in tuning indicator. Principle of these messages is described in next **section 10.5** and example of some message see in **fig. 10.15**.

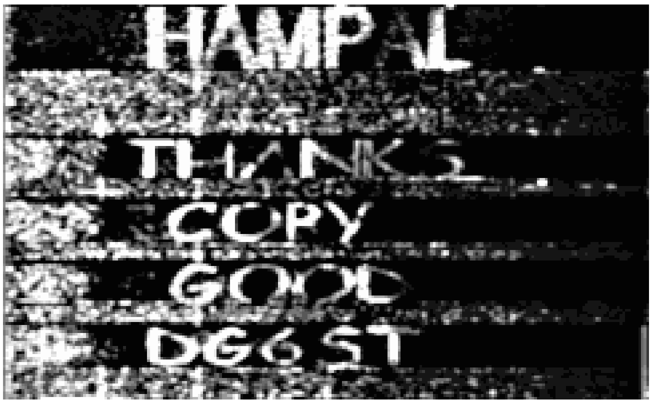


Figure 10.15: The confirmation of successful reception displayed in tuning indicator

The reports are same as for the phone operation in the RS (readability and strength) code. The V (View) value representing image quality of digital transmission is losing its importance. Readability is measured on a scale of 1 to 5, so level 5 stands for for a perfect error-free transfer, level 4 is 4 still acceptable, with occasional failure segments and potentially it's needed to increase the number of instances. Report the worst level 1 if can not receive any digital data.

Contrary to popular SSTV operations when stations restrict only to the exchange of images, the phone mode is much more used in case of DSSTV.

The choice of images is not limited to the usual 320×240 resolution, but there can be used any resolution. The limiting factor is only time of transmission, e.g. in the DIGTRX software the broadcast time is already known, so you can play around with the compression level, resolution, or number of colors and achieve a reasonable compromise.

Also, the transmitted data file format can be any. Listen on the band and you will make sure that JPEG2000 is often used, but also animated GIFs, or text files with ASCII art.

10.5 Waterfall images

For digital SSTV and RDFT or HamDRM system is used tuning indicator, which displays spectrum of SSB channel. The image showed by indicator is created using discrete Fourier transformation. The indicator displays new samples on top and the old samples disappear at the bottom and the whole spectrogram is moving down so the indicator was nicknamed *waterfall*.

In the **fig. 10.6** and **fig. 10.8** you can see station and software identification and also messages about reception confirmation, request for repeat or more complex pictures also.

The principle of "waterfall images" is based on Fourier transformation and the fact, that the signal can be compiled from a huge number of harmonic waves. If the proper harmonic are compiled, so the resulting carrier wave has frequency spectrum, that will look like desired image.

The utility `PicFall.exe` can be used for generating sound file from picture. You can find it on website of DIGTRX author. The input file is a bitmap in BMP format and output is WAV audio file.

Generate waterfall image by using `PicFall.exe`:
http://www.qsl.net/py4zbz/tutsstv14.htm

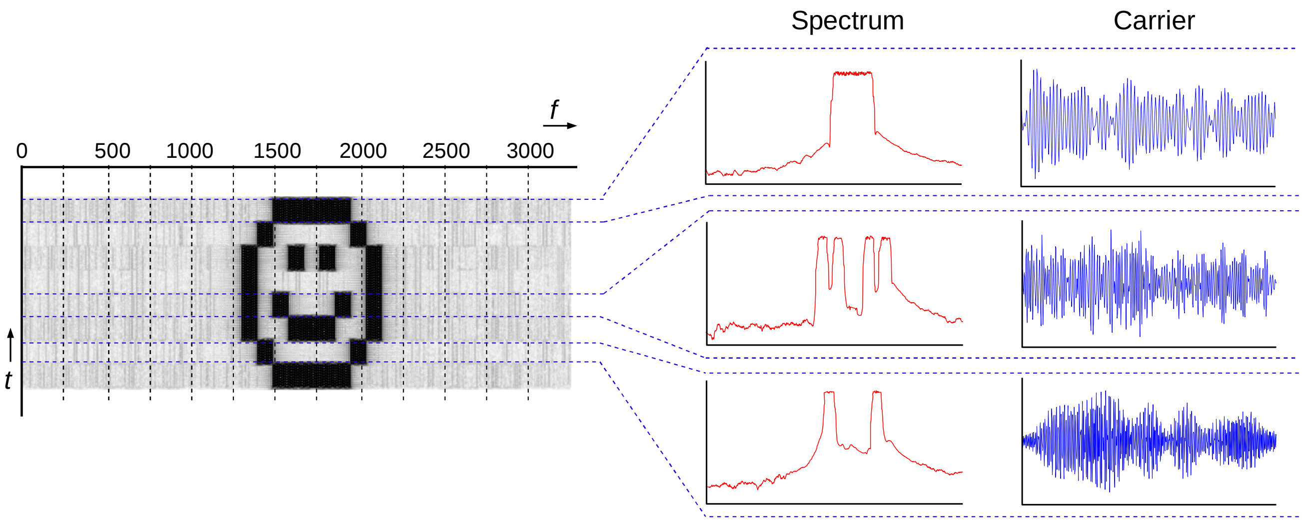


Figure 10.16: The principle of waterfall image display.