Products: DVRG, DVM100/400, SFU, EFA, DIP010, CMU200, CRTU-W, CRTU-G

Test of DVB-H-Capable Mobile Phones in Development and Production

Application Note

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ROHDE & SCHWARZ

Contents

1 Overview

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2 Introduction to DVB-H

The introduction of second-generation (2G) mobile radio (Global System for Mobile Communications/GSM) triggered a veritable boom in wireless communications. In the early 1990s, car telephones and similar mobile communications phones were largely reserved for special groups of users. But by the end of the decade, nearly every second person owned a mobile phone. Initially, mobile phones were primarily used to make phone calls or send and receive short messages (SMS). By the late 1990s, however, there was also a growing demand for using mobile phones to send and receive data, e.g. to or from a PC. Being able to check e-mail was a convenient way of staying up to date, especially for professional users on business trips. Today, this has more or less become standard. However, the data rates for the GSM standard, which was designed primarily for mobile phone communications, are in the order of 9600 bit/s. This is quite sufficient for simple text e-mails without any attachments. If large files are attached, however, this data rate is too low. Browsing the Internet is also possible using this standard, but doing so is expensive and tedious. With the introduction of 2.5G mobile phones (General Packet Radio System/GPRS), the available data rate was increased to 171.2 kbit/s by means of packetizing, i.e. by combining GSM timeslots. But it was only the third generation of mobile radio – the Universal Mobile Telecommunication System (UMTS) standard – that brought about a significant increase in data rate now measuring 144 kbit/s up to 384 kbit/s or 2 Mbit/s, where the actual data rate largely depends on the conditions surrounding reception or coverage in each case. The Enhanced Data Rates for GSM Evolution (EDGE) standard with higher-order modulation (8PSK) also makes it possible to achieve higher data rates of up to 345.6 kbit/s (ECSD) or 473.6 kbit/s (EGPRS).

All mobile radio standards are designed for bidirectional communication between the mobile terminal and the base station. The modulation methods – e.g. *Gaussian minimum shift keying* (GMSK) in the case of GSM, or *wideband code division multiple access* (WCDMA) in the case of UMTS – as well as the error protection method employed *(forward error correction*/FEC) are designed to withstand the adverse conditions of mobile reception.

Mobile phones long ago ceased to be just phones; they can now be used as cameras, playstations or organizers. Mobile phones are increasingly becoming multimedia terminals. Equipment manufacturers and network operators are continuously in search of new applications.

At the same time mobile radio was being developed, television migrated from analog to digital. At the end of the 1980s, the digital transmission of moving pictures over existing channels such as satellite, cable or the conventional terrestrial path still seemed impossible. Today, however, this is standard. Digital TV transmission has become possible through state-of-the-art compression techniques such as the Moving Pictures Expert Group (MPEG) technique, modern modulation methods and error protection (FEC) that matches the conditions at hand. The initial use of *discrete cosine transform* (DCT) in the Joint Photographics Expert Group (JPEG) standard can be seen as a key event in this field. JPEG is a method of compressing still pictures that is implemented in digital cameras. At the beginning of the 1990s, the experience gained with DCT was also applied to the compression of moving pictures in accordance with the MPEG standard. The initial standard – MPEG-1 – was designed to handle CD data rates and applications. MPEG-2 made it possible to compress *standard definition television* (SDTV) moving picture signals from data rates of initially 270 Mbit/s to below 5 Mbit/s, with the data rate of the associated lip-synchronous audio channel ranging from 200 kbit/s to 400 kbit/s. Even *high definition television* (HDTV) signals could be compressed to tolerable data rates of approx. 15 Mbit/s. Taking into account the special anatomical characteristics of the human eye and ear, a method referred to as *irrelevance reduction* was applied in combination with *redundancy reduction*. Signal content and information not perceived by the human eye or ear are removed from the signal prior to transmission.

Compression techniques have been continuously improved within the framework of MPEG-4 (H.264/MPEG-4 Part 10 AVC (Advanced Video Coding)). This standard makes even lower data rates possible while offering enhanced video and audio quality.

The development of Digital Video Broadcasting (DVB) has produced three new transmission modes: DVB-S (satellite), DVB-C (cable) and DVB-T (terrestrial). Of these, the DVB-S standard has virtually been fully implemented and enjoys broad acceptance among end users. DVB-S receivers are available at very favorable prices (starting from approx. 40 euros), with picture quality comparable to that of DVDs. The video and audio information on DVDs, by the way, is also MPEG-2-compressed. Digital TV is also supplied via many cable networks using DVB-C.

Today, a net data rate of approx. 38 Mbit/s is usually achieved in a satellite channel (transponder) or a cable channel. Satellite transponders operate in a bandwidth of approx. 33 MHz. Available bandwidths for cable TV channels are 6 MHz, 7 MHz or 8 MHz, where 8 MHz is nearly always used in practice. The use of Digital Video Broadcasting – Terrestrial (DVB-T) is also becoming more widespread on a global scale. In DVB-T, digital TV content is broadcast via terrestrial channels of 6 MHz, 7 MHz or 8 MHz. These broadcast channels are distributed over a frequency range from approx. 47 MHz to 862 MHz, with gaps in between. Typical net data rates are 15 Mbit/s or 22 Mbit/s. In some countries such as Great Britain, Sweden or Australia, DVB-T is intended for reception via a rooftop antenna only. This enables a relatively high data rate of approx. 22 Mbit/s. Other countries, e.g. Germany, opted for the "portable indoor" application. In densely populated areas, this means that over 20 services can be received free-to-air via a room antenna (passive or active). The second option requires more in-depth error protection (FEC) and a more robust modulation technique (16QAM instead of 64QAM). This reduces available data rates to, for example, approx. 15 Mbit/s. DVB-T receivers are meanwhile available at reasonable prices from approx. 80 euros. Portable solutions in the form of PCMCIA cards are available for laptops. In cable and satellite transmission, a modulation method with a single carrier is

used. DVB-S employs QPSK with symbol rates of usually 27.5 Msymb/s; DVB-C often uses 64QAM or 256QAM with symbol rates of approx. 6.9 Msymb/s. DVB-S and DVB-C channels are capable of transporting a large number of services. For network operators, DVB-S and DVB-C offer clear economic advantages over conventional analog transmission while providing better video quality.

In DVB-T networks that provide portable reception, data rates are in the order of 15 Mbit/s. As a result, only about four services can be transported via a DVB-T channel. It should be noted, however, that this is four times the volume previously transported over a comparable analog TV channel. The data rates for the individual services are thus in the order of 2.5 Mbit/s to 3.5 Mbit/s; these are usually variable data rates forming part of a statistical multiplex.

2.1 Convergence of Mobile Radio and Broadcasting

Mobile radio networks allow robust, bidirectional (point-to-point) connections using relatively low data rates. Modulation methods, error protection levels and handover techniques are matched to the more difficult conditions of mobile transmission. Consequently, billing and similar features form an integral part of the system. The end user selects the services to be provided – for example, a phone call, an SMS or a data link. Charges are calculated on the basis of the services requested.

Broadcast networks are unidirectional networks involving the point-to-multipoint distribution of identical content to a large number of subscribers at relatively high data rates. Content-on-demand is relatively rare; instead, predefined content is distributed to a large number of subscribers from a transmitter location or, more recently, through *single-frequency networks* (SFN). The content is usually a radio or TV service. Data rates are significantly higher than in mobile radio networks. The modulation methods and error protection levels used are often only designed for portable reception or reception via a rooftop antenna. Only the Digital Audio Broadcasting (DAB) standard provides for mobile reception. DVB-T, by contrast, was developed for stationary and portable reception only.

The purpose of the new DVB-H standard is to merge mobile radio and broadcasting in order to offer the advantages of both systems. This involves combining bidirectional mobile radio networks, which rely on relatively low data rates, with unidirectional broadcast networks, which use relatively high data rates. If numerous subscribers then request the same service or services at the same time, e.g. a specific video or audio service on demand, the mobile network's data service will be mapped to the broadcast network point-to-multipoint. This will be done depending on requirements and the amount of content to be transported.

The exact content to be redirected from the mobile radio network to the broadcast network will be determined solely by the current requirements in each case. To date, no decision has been made as to the services to be provided to mobile phone subscribers via DVB-H. Purely IP-based services or video/audio via IP may be offered. In any case, DVB-H will be implemented as a UDP/IP-based service as part of MPEG/DVB-T/H. Possible applications include live transmissions of sports events, news programs and other services that may be of interest to a large number of mobile subscribers. Mobile phones compatible with the DVB-H standard will certainly also be capable of receiving free DVB-T broadcasts, depending on current reception conditions.
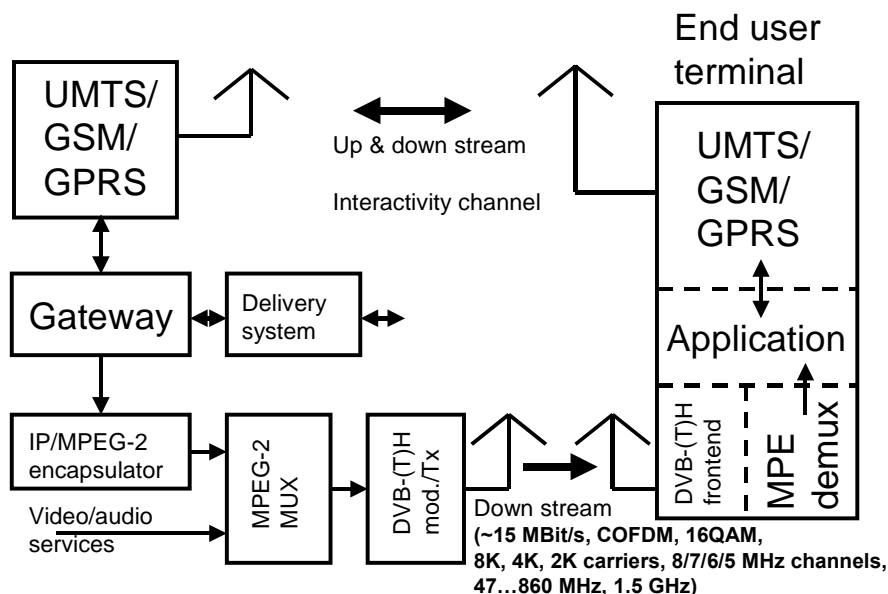
Fig. 1: Convergence of mobile radio an DVB

2.2 Main DVB-H Parameters

The main DVB-H parameters conform to the DVB-T standard. Only slight expansions have been made to the DVB-T physical layer. In addition to the 8k and 2k modes already available in DVB-T, the 4k mode has been introduced as a good compromise between the two. It is more compatible with mobile phones and also allows SFNs of reasonable size to be created. The 8k mode is not well-suited for mobile phones because of its low subcarrier spacing, while the 2k mode only allows short transmitter distances of approx. 20 km. The 8k mode requires more memory for interleaving and de-interleaving data than does the 4k or 2k mode. Memory that is freed up in the 4k and 2k modes can be used for more in-depth interleaving in DVB-H. This means that the interleaver can select between native and in-depth interleaving in the 4k and 2k modes. To make it possible to signal additional parameters, transmission parameter signaling (TPS) bits that are reserved or even already in use elsewhere are implemented in DVB-H.

The additional parameters introduced with DVB-H are included in the DVB-T standard ETS 300744 (ETSI, European Standardization Institute) as an appendix. All further changes or expansions are specific to the MPEG-2 transport stream; they are provided in the DVB Data Broadcast Standard ETS 301192. As a DVB baseband signal, the MPEG-2 transport stream is the input signal for a DVB-H modulator. Multiprotocol encapsulation (MPE), which had already been defined in DVB data broadcasting prior to DVB-H, is implemented with time slicing in DVB-H in order to save power in the mobile phone. Both the timeslot length and spacing must be signaled. In DVB-H, the IP packets packed in the MPE timeslots can be furnished with additional FEC if necessary. FEC is a Reed-Solomon error protection mechanism at the IP packet level. Everything else corresponds directly to

DVB-T or MPEG-2. Thus, the following text also initially describes the MPEG-2 transport stream and the DVB-T standard so that the main differences between DVB-H and DVB-T can subsequently be addressed.

Summary: DVB-H is a method of transmitting IP packets via an MPEG-2 transport stream by using time slicing. The physical layer is actually DVB-T with a few expansions. The objective is to converge a mobile radio network and a DVB-H broadcast network. Depending on the amount of data involved, data services are transmitted to the mobile phone either via the mobile radio network or via the DVB-H network.

2.3 Basic Principles of DVB

The input signal to a DVB modulator is an MPEG-2 transport stream. The structure of this stream is defined by the ISO/IEC 13818-1 standard, "MPEG-2 Systems". An MPEG-2 transport stream, abbreviated MPEG-2 TS, is made up of short, 188-byte packets referred to as *transport stream (TS) packets*. Each packet starts with a four-byte header, which is followed by 184 bytes of payload. The first byte of the header is the sync byte. It has a constant value of 47 hex and occurs every 188 bytes at the beginning of each TS packet. The byte is needed for synchronization with the MPEG-2 data signal. For synchronization, the 47 hex value is checked every 188 bytes. The value of 47 hex may also occur in any other section of the TS packet. The TS header also contains the packet identifier (PID). This is a 13-bit value from which the content currently being transported in the payload component of the TS packet can be determined.
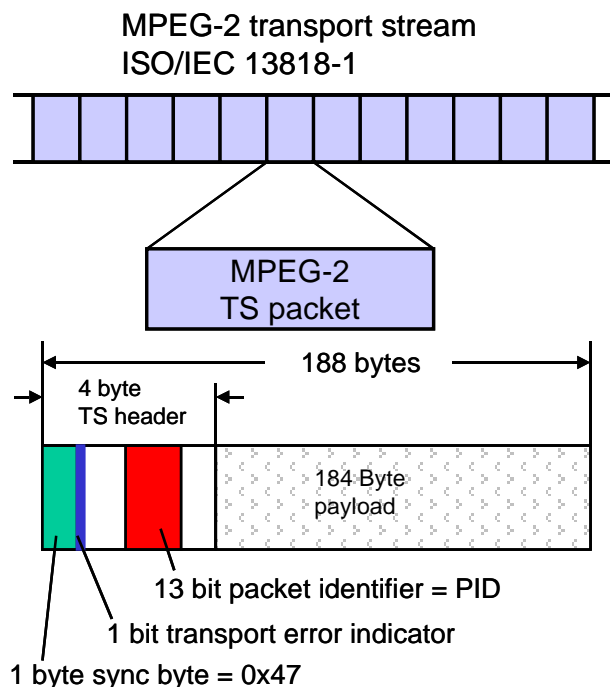

Fig. 2: MPEG-2 transport stream

The transport stream also includes tables that describe the assignment of PIDs to useful content such as video, audio and data. The information provided by the tables is referred to as *program specific information/service information* (PSI/SI). Initial information is provided by the *program*

association table (PAT), which can be recognized by the PID "0". The PAT contains a PID entry for each service (or program). The PID points to a *program map table* (PMT), which exists for each service. Each PMT, in turn, includes PIDs as pointers to the associated content such as video, audio and data. Video and audio signals are present in the form of elementary streams with special PIDs. The elementary streams are compressed in line with MPEG. They are also organized in packets and are therefore referred to as *packetized elementary streams* (PES). PES packets may be up to 64 kbytes in length and likewise consist of a header and a payload component.
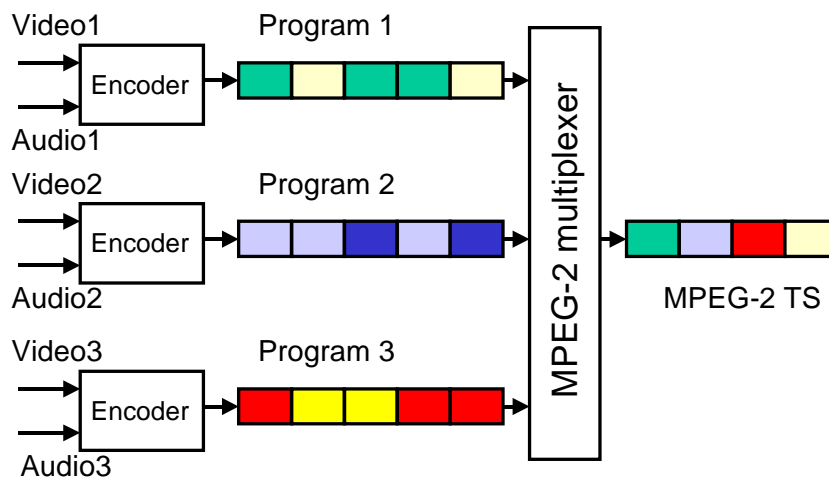

Fig. 3: MPEG-2 multiplexing

In compliance with MPEG, a special feature of packetized elementary streams is the capability to synchronize with one another, which is of particular importance in lip synchronization between video and audio, for example. Packetized video and audio elementary streams are generated in the MPEG encoder by compressing the video and audio signals. The PES packets of up to 64 kbytes in length are divided into TS packets before they leave the MPEG encoder, and are multiplexed as video and audio packets. The MPEG multiplexer then multiplexes, i.e. time-interleaves, several services to produce a transport stream. The transport stream that leaves the multiplexer is constructed of TS packets that carry the content of various services. The content can be identified from the different PIDs, and solely from these PIDs.

The PSI/SI tables inform the DVB receiver of the elements contained in the multiplex stream and supply further useful information such as the service name or even an electronic program guide. Moreover, the date and time are transmitted. The corresponding tables are the PAT, PMT, CAT, NIT, SDT, BAT, EIT, RST, TDT and TOT. The functions of these tables are listed below. For details, refer to the relevant literature ["Digital Television, A Practical Guide for Engineers" by Walter Fischer, published by Springer in 2004].

The PSI/SI tables are transmitted in sections in the transport stream. Sections are data packets of up to four kbytes in length whose structure is defined in the MPEG-2 system layer. They are divided in 184-byte segments and transmitted in the payload component of TS packets that are assigned special PIDs. Each section starts with a table ID. This is a byte that, in conjunction with the PID of the TS packet in which it is transmitted, defines the table type. The table ID is followed by the header of the section. Several sections together yield a table.
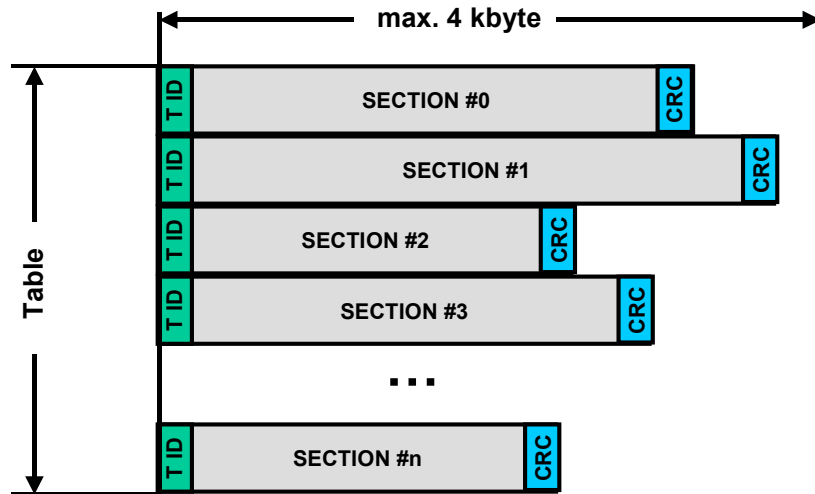

Fig. 4: Sections and tables

A DVB-compliant MPEG-2 transport stream thus contains elementary streams that can be synchronized with one another and that transmit services with video, audio and even teletext. In other words, one transport stream carries multiple services. Moreover, information about the transport stream structure as well as user information is transmitted from the multiplexer to the DVB receiver in sections that contain PSI/SI tables.

PSI/SI table			PID	Table ID
PAT	Program association table		0x0000	0x00
PMT	Program map table		0x0020..0x1FFFE	0x02
CAT	Conditional access table		0x0001	0x01
NIT	Network information table		0x0010	0x40…0x41
SDT	Service descriptor table		0x0011	0x42, 0x46
BAT	Bouquet association table		0x0011	0x4A
EIT	Event information table		0x0012	0x4E…0x6F
RST	Running status table		0x0013	0x71
TDT	Time & date table		0x0014	0x70
TOT	Time offset table		0x0014	0x73
ST	Stuffing table		0x0010…0x0014	0x72

Table 1: PSI/SI tables

2.4 DSM-CC Sections

Part 6 of the ISO/IEC 13818 MPEG-2 standard specified mechanisms for transmitting data, data services and directory structures early on. These mechanisms are known as *data storage media command and control* (DSM-CC) sections. The structure of DSM-CC sections is basically comparable to that of PSI/SI tables. The sections start with a table ID that is always between 0x3A and 0x3E. DSM-CC sections are up to four kbytes in length, and their content is likewise distributed among a number of TS packets, which are multiplexed into the transport stream and broadcast. Using object carousels (which broadcast data in repeating cycles), complete directory trees with various files can be transported to the DVB receiver via DSM-CC sections. This can be done, for example, by means of a *multimedia home platform* (MHP), which is used to transmit HTML and Java files that can then be executed on MHP-compatible DVB receivers.

DSM-CC sections with a table ID of 0x3E can transmit IP packets in the MPEG-2 transport stream. Each IP packet transports either a Transmission Control Protocol (TCP) packet or a User Datagram Protocol (UDP) packet. TCP packets use a handshake to perform a controlled transmission between a transmitter and a receiver. In contrast, UDP packets are transmitted without any acknowledgment. Broadcasting, as the term implies, normally does not involve a back channel, i.e. TCP packets do not come into play. For this reason, only UDP protocols are used in MPE, which is employed for transmitting IP packets in DVB. Although the mobile radio network would provide a back channel in DVB-H, it is not possible to prompt for a new IP packet since the messages must be sent to many destinations simultaneously.
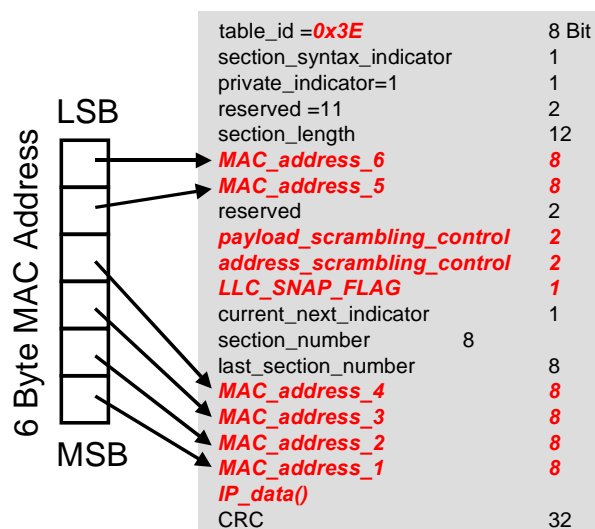
table_id = *0x3E*	8 Bit
section_syntax_indicator	1
private_indicator=1	1
reserved =11	2
section_length	12
MAC_address_6	*8*
MAC_address_5	*8*
reserved	2
payload_scrambling_control	*2*
address_scrambling_control	*2*
LLC_SNAP_FLAG	*1*
current_next_indicator	1
section_number	8
last_section_number	8
MAC_address_4	*8*
MAC_address_3	*8*
MAC_address_2	*8*
MAC_address_1	*8*
IP_data()	
CRC	32

Fig. 5: DSM-CC datagram section

2.5 Multiprotocol Encapsulation (MPE)

MPE is used in DVB to transport content such as HTML files or MPEG-4 video and audio streams in UDP packets. Windows Media 9 applications can also be broadcast and replayed on suitably equipped terminals. Each UDP packet contains a destination port address (DST port), which is a 16-bit numeric value used to address the destination application. The World Wide Web (WWW), for example, uses 80 hex as the port number for all of its communications. Ports are protected and access-controlled by means of firewalls.

UDP packets are embedded in the payload component of IP packets. The IP packet header contains the source and destination IP addresses (SRC and DST IP addresses) necessary for routing the IP packet through a network from a transmitter to a receiver.

In a typical computer network, IP packets are usually transported within Ethernet packets. In this case, the Ethernet packet header contains the hardware addresses of the intercommunicating network components; these addresses are known as media access command (MAC) addresses.

When IP packets are to be transmitted via DVB networks, the Ethernet layer is replaced by the MPEG-2 transport stream and the physical DVB layer (DVB-C, -S, -T). The IP packets are packaged in DSM-CC sections, which are then divided into multiple TS packets. Thus, MPE consists of placing UDP packets into IP packets, IP packets into DSM-CC sections, and DSM-CC sections into TS packets. Here, the header of the DSM-CC section contains the *destination MAC* (DST MAC) address. As in the case of the Ethernet layer, this address is six bytes in length. A source MAC address is not included. For details, refer to ISO/IEC 13818-6, ETS 301 192 DVB Data Broadcasting.
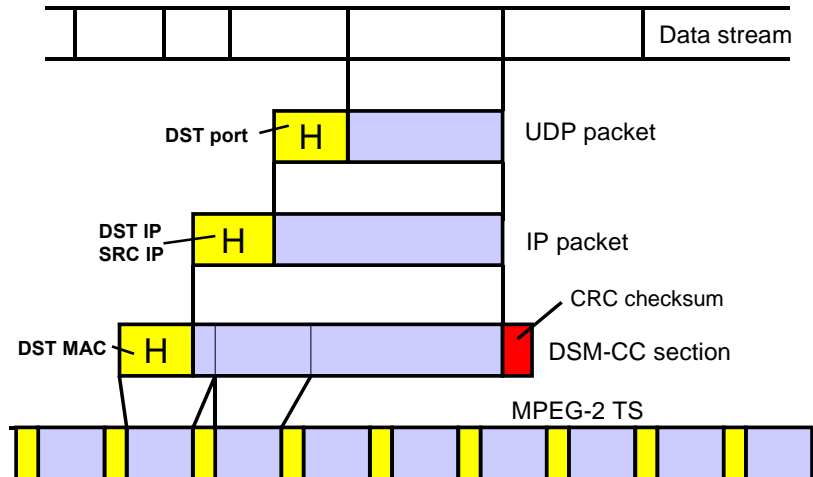
Fig. 6: Multiprotocol encapsulation (MPE)

2.6 DVB-T: Digital Video Broadcasting – Terrestrial

The input signal to each DVB modulator, whether DVB-C, DVB-S, DVB-T or DVB-H, is an MPEG-2 transport stream. In the case of DVB-T, a modulation mode that can handle the difficult conditions of terrestrial transmission is employed, i.e. *coded orthogonal frequency-division multiplex* (COFDM). Terrestrial transmission is characterized by multipath reception (caused by echoes from mountains, trees, buildings, etc), interference, noise and – in mobile reception – by frequency shifts resulting from the Doppler effect. Multipath reception leads to fading at selective frequencies and locations; this means that the signal is weakened or even canceled out at certain points in the spectrum. Apart from the effect in the frequency domain, multipath reception also has an effect in the time domain, namely *intersymbol interference* (ISI). Neighboring symbols overlap, causing noise-like impairments in digital transmissions.
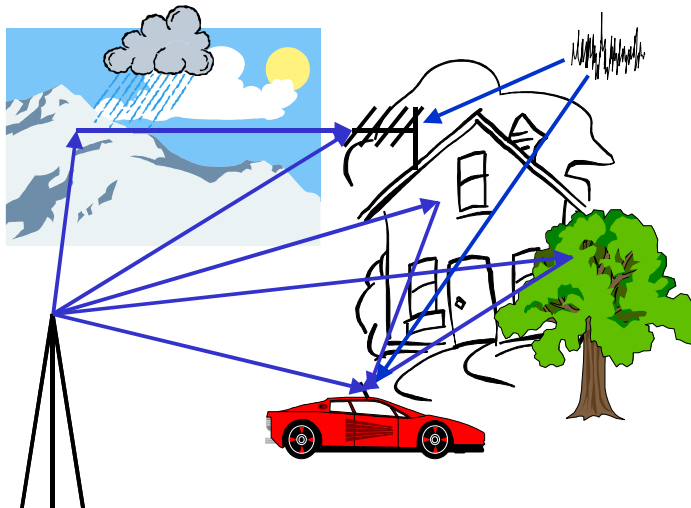

Fig. 7: Terrestrial transmission channel

Instead of a single carrier, COFDM modulation may use several thousands of narrow-bandwidth carriers in one channel. The data stream undergoes error protection before it is fed to the OFDM modulator. Each COFDM carrier transmits part of the error-protected data stream. Individual subcarriers that are affected by coverage gaps usually cannot be demodulated. With the data signal distributed among many subcarriers, however, the overall data stream can be retrieved.

COFDM modulation involves very long symbol durations (up to 1 ms in DVB-T) since the information to be transmitted is distributed among many subcarriers. This enables guard intervals to be inserted between symbols without any significant reduction in the useful data rate. The guard interval provides a means of avoiding ISI. Thus, COFDM, which features error protection as well as the distribution of information to multiple carriers in the transmission channel, makes it possible to handle fading effects in the frequency domain and ISI in the time domain.
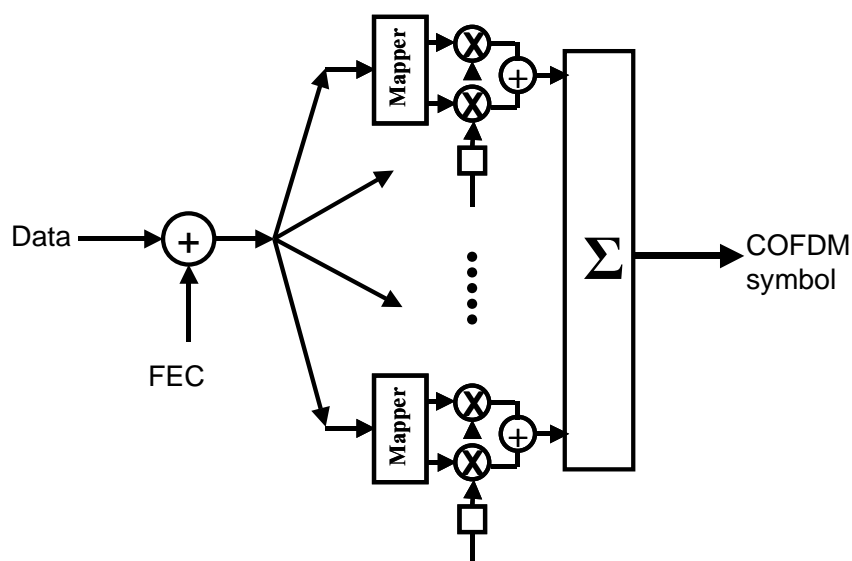

Fig. 8: COFDM modulator

In DVB-T, a 2k and an 8k mode are available. In the 2k mode, 2048 carriers are processed, hence its name. Only 1705 carriers are actually used, however. The remaining carriers form a guard band extending up to the adjacent channels. In the 8k mode, 8192 carriers are processed. Of these, 6817 carriers are actually used; the remaining carriers form the guard band. The 2k mode involves relatively short symbols of approx. 250 µs, the 8k mode relatively long symbols of approx. 1 ms. The guard interval in DVB-T can be set to 1/4, 1/8, 1/16 or 1/32 of the symbol duration. The longest possible guard interval is therefore approx. 250 µs, corresponding to 1/4 of the symbol duration in the 8k mode. During the guard interval, the DVB-T signal does not pause; instead, the end of the subsequent symbol is repeated. This makes it easier for the receiver to synchronize.

A 250 µs guard interval involves a maximum echo delay corresponding to approx. 70 km. In the 2k mode, by contrast, the maximum echo delay is only 1/4 of the delay encountered in the 8k mode, corresponding to

somewhat less than 20 km. The 2k mode, on the other hand, is clearly more robust when it comes to handling smearing effects in the spectrum, which occur under conditions of combined mobile and multipath reception. This application note will later discuss how an additional mode, the 4k mode, has been introduced in DVB-H as a compromise.
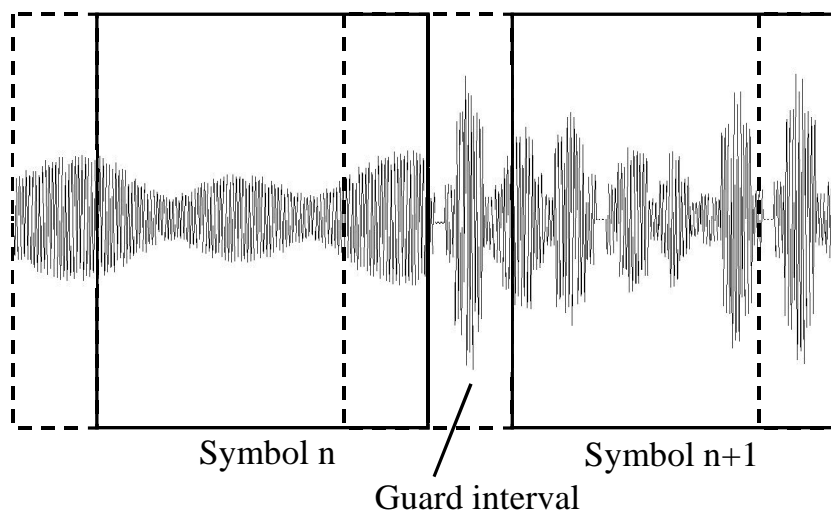

Fig. 9: COFDM symbol with guard interval

In DVB-T, FEC is implemented by means of Reed-Solomon block coding concatenated with convolutional coding. As a first step, the MPEG-2 transport stream is scrambled in the energy dispersal block in order to break up long sequences of zeros or ones. This yields a data stream with a relatively random structure. The sync bytes in the data stream remain unscrambled as they are required for the receiver to synchronize with the signal. The next step is Reed-Solomon coding, where 16 error control bytes are added to each 188-byte TS packet. With the aid of these 16 bytes, the receiver can correct up to eight errored bytes of a TS packet, which is now 204 bytes in length. This is followed by a convolutional interleaver, which rearranges the chronological sequence of data bytes, thus spreading information over a longer period of time. In the process of de-interleaving at the receiver end, burst errors are then broken down into single errors, which are easier to correct for the Reed-Solomon decoder. The convolutional interleaver is followed by a convolutional coder, which expands the data stream by a factor of two. This overhead can be reduced by means of subsequent puncturing. To do this, a code rate is applied, which constitutes the ratio of the input data rate at the convolutional encoder to the output data rate of the puncturing stage. Code rates of 1/2, 2/3, 3/4, 5/6 and 7/8 are selectable. A code rate of 1/2 means maximum error protection, a code rate of 7/8 minimum error protection.

The error-protected data stream then undergoes bit and symbol interleaving, where information is distributed among the individual payload carriers (6048 in the 8k, 1512 in the 2k mode) across the available frequency range. Bits are then mapped to symbols by which the COFDM carriers are vector-modulated. Either QPSK, 16QAM or 64QAM can be used. In the frame adaptation block, continual pilots, scattered pilots and transmission parameter signaling (TPS) pilots are inserted at specific reserved positions in the spectrum. The continual pilots are inserted at fixed

carrier positions; they are used for *automatic frequency control* (AFC). The scattered pilots are quasi-sweep signals and are used for channel estimation and channel correction. The TPS pilots or carriers form a fast information channel in which the DVB-T modulator uses 68 COFDM symbols to inform the DVB-T receiver about the current DVB-T transmission parameters.

DVB-T system parameters include:

- 8k/2k mode
- QPSK, 16QAM, 64QAM
- Hierarchical modulation on/off
- Code rate 1/4 to 7/8
- Guard interval 1/4 to 1/32
- Cell ID

This application note will also discuss how changes or expansions have been made with DVB-H in order to broadcast additional information. In all, 67 bits are transmitted via 68 COFDM symbols. The bits are modulated on the TPS carriers using DBPSK modulation.

After the mapping stage and the insertion of the pilots, the COFDM symbols are generated by means of an *inverse fast Fourier transform* (IFFT). Then the guard intervals are inserted, which may have a length of 1/4, 1/8, 1/16 or 1/32 of the symbol duration. The COFDM symbols first undergo precorrection and are then upconverted to the RF by direct modulation in the IQ modulator, applied to an amplifier stage and transmitted via an antenna.

2.7 The DVB-H Standard

The purpose of the DVB-H standard is to converge mobile radio networks and broadcast networks. If traffic volume is large enough, the downstream is mapped from a mobile radio network (GSM/GPRS, UMTS) to a broadcast network. For example, if only one subscriber requests a service via UMTS, this service will continue to be provided via UMTS in the downstream. However, if numerous subscribers request the same service at the same time, it is worthwhile to offer this service (a video, for example) point-to-multipoint via a broadcast network. The services to be implemented via DVB-H are IP-packet-based throughout.

As to the physical implementation of the DVB-H standard, a modified DVB-T network is to be used to broadcast IP services in timeslots in an MPEG-2 transport stream. The physical modulation parameters of the DVB-H network are very similar, and in part identical, to those of a DVB-T network. More significant changes are required to the MPEG-2 transport stream, however.

The ETSI document TM 2939 provides an overview of the DVB-H system. A detailed description can be found in the DVB Data Broadcasting Standard ETS 301 192 and in the DVB-T Standard ETS 300 744. The main parameters are also presented in this application note.

Least affected, and thus requiring the least degree of modification, is the physical DVB-T layer. An optional 4k mode was introduced as a compromise between the 8k mode, which is particularly suitable for SFNs, and the 2k mode, which is better suited for mobile reception. Using the 4k mode doubles transmitter spacing in SFNs as compared with the 2k mode, and substantially improves mobile reception as compared with the 8k mode. Storage space that is freed up in the interleaver and the de-interleaver through the use of the 4k or 2k mode is to be utilized for in-depth interleaving. The purpose of in-depth interleaving is to make DVB-H more robust against burst errors, i.e. multiple bit errors. The time distribution of the data stream is improved as a result.

| **2K Mode** $\Delta f \sim 4kHz$, $t_s \sim 250us$	**4K Mode** $\Delta f \sim 2kHz$, $t_s \sim 500us$	**8K Mode** $\Delta f \sim 1kHz$, $t_s \sim 1000us$
2048 carriers	4096 carriers	8192 carriers
1705 used carrier	3409 used carrier	6817 used carrier
Continual pilots	Continual pilots	Continual pilots
Scattered pilots	Scattered pilots	Scattered pilots
TPS carrier	TPS carrier	TPS carrier
1512 data carrier	3024 data carrier	6048 data carrier
In-depth inter-leaving on/off	In-depth inter-leaving on/off	

Fig. 10: Overview of 2k, 4k and 8k modes

In the DVB-H standard, a number of additional system parameters have to be signaled via TPS pilots.

These parameters are:

- Time slicing on/off in the MPEG-2 transport stream (i.e. DVB-H transmission on/off)
- IP packet transmission with FEC on/off
- In-depth interleaving on/off
- 4k mode

DVB-H parameter signaling is achieved by using two further bits (bits 42 and 43) of the reserved TPS bits as well as TPS bits already in use. Details are shown in the figure below.
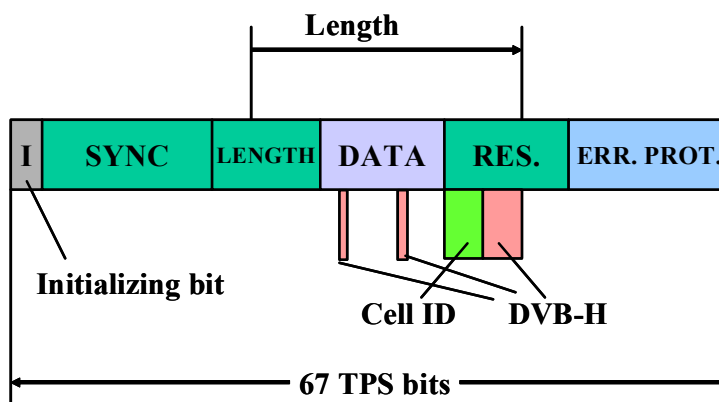
Bit 27…29 = Hierarchical Mode 000, 001, 010, 011

Bit 27: 0 = native interleaver

       1= depth interleaver (only in 2K and 4K mode)

Bit 38, 39: *00 = 2K, 01=8K*, 10=4K mode

Bit 40…47 = cell ID

2 completely new TPS bits in use:

Bit 48: DVB-H (time slicing) ON/OFF

Bit 49: IP FEC ON/OFF

Fig. 11: TPS bits

RF performance in the mobile channel can be improved by using the 4k mode and in-depth interleaving in the 4k or 2k mode. In addition, transmitter spacing in an SFN network can be increased by a factor of two in the 4k mode (approx. 35 km) as compared with the 2k mode (approx. 17 km).

In addition to the channel bandwidths of 8 MHz, 7 MHz and 6 MHz, which are standard in DVB-T, a bandwidth of 5 MHz (L band, USA) will also be available in DVB-H.

More substantial modifications have to be made to the structure of the MPEG-2 transport stream.

In DVB-H, IP packets are transmitted in an MPEG-2 transport stream using MPE (see above). MPE in DVB-H has a few special features compared with MPE as employed in DVB-T, for example. IP packets can be protected by means of an additional Reed-Solomon FEC. The Reed-Solomon FEC of an IP datagram is broadcast in separate MPE-FEC sections. These sections have a table ID of 0x78. The header of these FEC sections has the same structure as that of the MPE sections. Since the FEC is transmitted separately, a receiver can retrieve the IP packet even without performing FEC analysis if no errors have occurred. Moreover, the IP packet information to be transmitted is bundled in timeslots in the MPEG-2 transport stream, a procedure referred to as *time slicing*. The DSM-CC header in the timeslots carries information that signals the time Δt until the start of the next timeslot. Thus, after receiving a timeslot, the mobile phone can switch to sleep mode until shortly before the next timeslot is received in order to save battery power. Fig. 12 shows the actual and target power consumption. Data rates in the timeslots will be up to approx. 400 kbit/s on average, depending on the application. The transmitted content is IP packet information that is requested by many subscribers at the same time. To signal the time Δt until the next timeslot, four out of the six bytes allocated to the destination MAC address in the DSM-CC header are used. The end of a timeslot is signaled via the frame boundary and table boundary bits in the MPE and FEC sections. A new SI table, the *IP MAC notification table* (INT) has been introduced in the MPEG-2 transport stream. This table informs the mobile receiver where to find a requested IP service; in addition, the table carries the timeslot parameters.
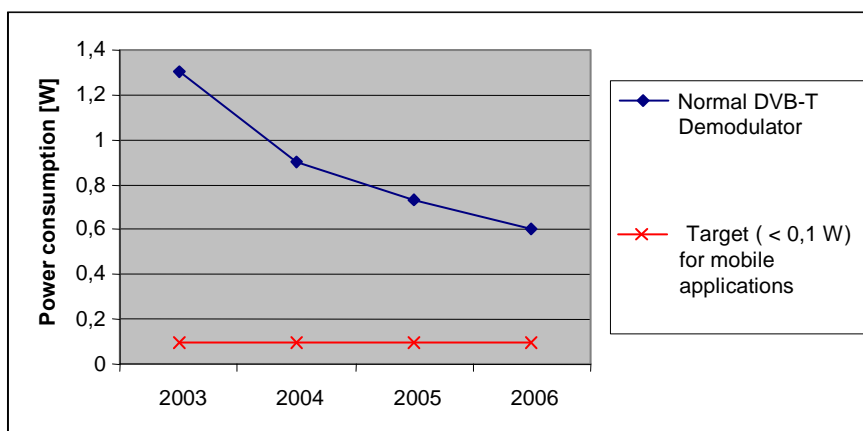

Fig. 12: Power consumption of DVB-T and DVB-H receivers

2.8 Summary

DVB-H is a combination of the GSM/UMTS and the DVB standards. The GSM/UMTS mobile radio network is used as an interactive channel. This network enables subscribers to request services involving higher data rates, e.g. video streaming (H.264/ MPEG-4 Part 10 AVC (Advanced Video Coding) or Windows Media 9). The services are made available to subscribers via the mobile radio network (UMTS) or via the DVB-H network. DVB-H basically uses a slightly modified DVB-T network. Additional operating modes, or transmission parameters, have been introduced in DVB-H. These include:
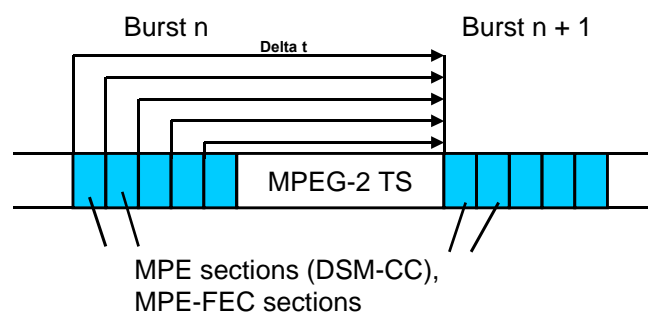
- The 4k mode with 3409 carriers for useful data as a viable compromise between the 2k and 8k modes
- In-depth interleaving in the 4k mode and also in the 2k mode
- Signaling of additional (DVB-H) parameters via two new TPS bits and via TPS bits already in use
- Time slicing for reducing power consumption
- IP packets with FEC error protection
- Introduction of a 5 MHz channel (L band, USA)

Information is transmitted in the MPEG-2 transport stream using MPE in conjunction with time slicing. IP packets can be protected by means of an additional Reed-Solomon FEC. A new DVB SI table (IP MAC notification table/INT) has been introduced to inform the user terminal where to find a requested IP service.

An initial prototype of a DVB-H-compatible user terminal was presented at the end of 2003.

Now that the basic principles of DVB-H have been presented, the following sections will discuss how system parameters of DVB-H-compatible mobile phones are tested.
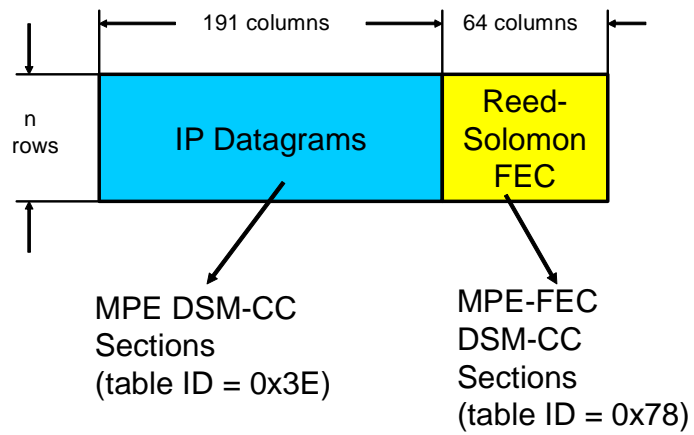

Fig. 13: Time slicing in DVB-H

Fig. 14: MPE and FEC sections

3 Mobile Radio Layer Testing

In the future, the various mobile broadcast standards – particularly DVB-H – will work together with the various mobile radio standards implemented throughout the world.

This section provides an overview of the test requirements for mobile radio systems in mobile phones.

A wide variety of mobile radio standards are in use worldwide. The most important cellular networks are GSM, (E) GPRS, CDMA2000 1x, 1xEV-DO, TDMA (IS136), AMPS, WCDMA and HSDPA.

Mobile phone tests vary depending on the stage of the product cycle, i.e. development (hardware/software), certification, production or service.

In addition to cellular networks, many mobile phones also already contain data transfer modules based on standards such as *Bluetooth*® und WLAN 802.11, and these modules must be tested at the same time as the mobile radio network. Like with the mobile broadcast standards, this has led to problems such as crosstalk or significantly greater demands on mobile phone batteries.

One area that is becoming particularly more important is the testing of handover scenarios with dual-band, triple-band and multimode mobile phones – between GSM/GPRS and WCDMA, for example. Handovers of this nature save substantial measurement time in production. Similar scenarios are also being considered between DVB-H and mobile radio networks (*convergence of broadcast and mobile services*/CBMS).

All components of a mobile phone must be tested. Fig. 15 shows a block diagram of a GSM mobile phone.

The following test steps need to be differentiated, particularly in production lines:

- Module test (board level test)

- Functional test

- RF adjustment

- Final test including RF, acoustic, keyboard and display tests

A typical functional test for GSM/GPRS mobile phones is presented here.

The first step is to perform general tests such as the following:

– Short-circuit test

– Measurement of the mobile phone's current drain, etc

The next step is to perform various measurements to verify that the transmitter is functioning properly:

– RF power

– Frequency errors
– Spectrum
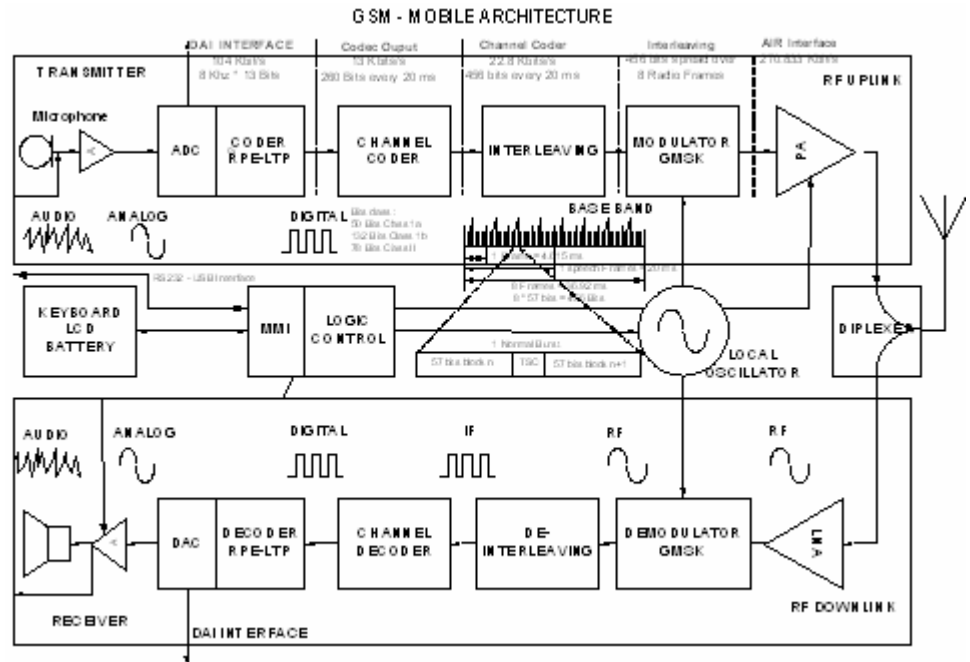– Modulation errors (magnitude error, phase error)

Fig. 15: Block diagram of a GSM mobile phone

The following items must be checked in receiver measurements:

– Automatic gain control (AGC)

– Received signal strength indicator (RSSI)

In addition, various bit error ratio (BER) measurements must be performed to ensure that the mobile phone has a sufficient level of sensitivity.

Next, measurements are performed on oscillators:

– AFC

– VCO calibration

The next step is to verify that the mobile phone's audio system is functioning properly by performing the following measurements:

– Audio frequency response

– Distortions

The final step is to test the digital section:

– Memory test

– SIM card interface

Test sequences that are becoming increasingly time-critical are being developed, particularly for RF adjustment (e.g. special calibration routines for level adjustment). Innovative modulator concepts (zero IF, polar modulator) demand new test strategies.

The introduction of data services such as the sending/receiving of multimedia message services (MMS), Internet browsing and video streaming are making application tests increasingly more important alongside the RF parameter measurements mentioned above.

Application test requirements will become even more rigorous as mobile broadcast standards such as DVB-H are introduced.

3.1 Overview of Test Instruments

Rohde & Schwarz offers an optimum portfolio of instruments for mobile radio tests:

Universal Radio Communication Tester R&S CMU 200

- Multiprotocol mobile station test platform supporting all major 2G/3G standards as well as *Bluetooth*
- Ready for future expansions such as HSDPA and CDMA2000 1xEV-DO Rel. A
- Only currently available solution that meets the requirements of acoustic measurements for 3GPP and 3GPP2
- Optimized for production and R&D; high-speed testing, excellent accuracy and extraordinary flexibility

Fig. 16: Universal Communication Tester CMU200

Protocol Tester R&S CRTU

- Protocol test system approved by the Global Certification Forum (GCF)
- Verification of protocol stack implementation
- 2G/3G intersystem handover testing
- Radio resource management verification
- Application and data testing
- Flexible programming interface for user-specific
- and test case scenarios

Fig. 17: Protocol Tester R&S CRTU-W for WCDMA/HSDPA

4 Generation of DVB-H Baseband Signals – Testing a DVB-H Receiver

4.1 Main Characteristics of DVB-H Transport Streams

The implementation of DVB-H requires significant transport stream expansions in comparison to conventional transport streams for TV-only applications.

The main expansions are as follows:

- Signaling of the DVB-H service (INT table, various descriptors)
- Coding of the useful data in accordance with application-dependent methods
- Transmission of useful data (IP packets) via MPE
- Addition of error correction information (MPE-FEC)
- Implementation of time slicing
-

4.2 Main Characteristics of Time Slicing and the MPE-FEC Frame

Fig. 16 provides an overview of the main parameters for time slicing and the MPE-FEC frame. Note that there are various dependencies between the specified parameters. For example, the equivalent bit rate (corresponding to the data rate of the DVB-H service) is derived from the burst bit rate and the quotient of burst duration to burst cycle time. The MPE-FEC frame parameters determine the amount of useful data in such a frame. Since exactly one complete frame is transmitted for each burst, there is a direct correlation between the equivalent bit rate, the MPE-FEC frame parameters and the burst duration.

The delta_t realtime parameter is transmitted with each section; it indicates the time until the start of the next burst. Thus, its value varies from section to section. Note that the actual receiver off-time is somewhat less than the theoretical value due to synchronization times and time reserved for other purposes.
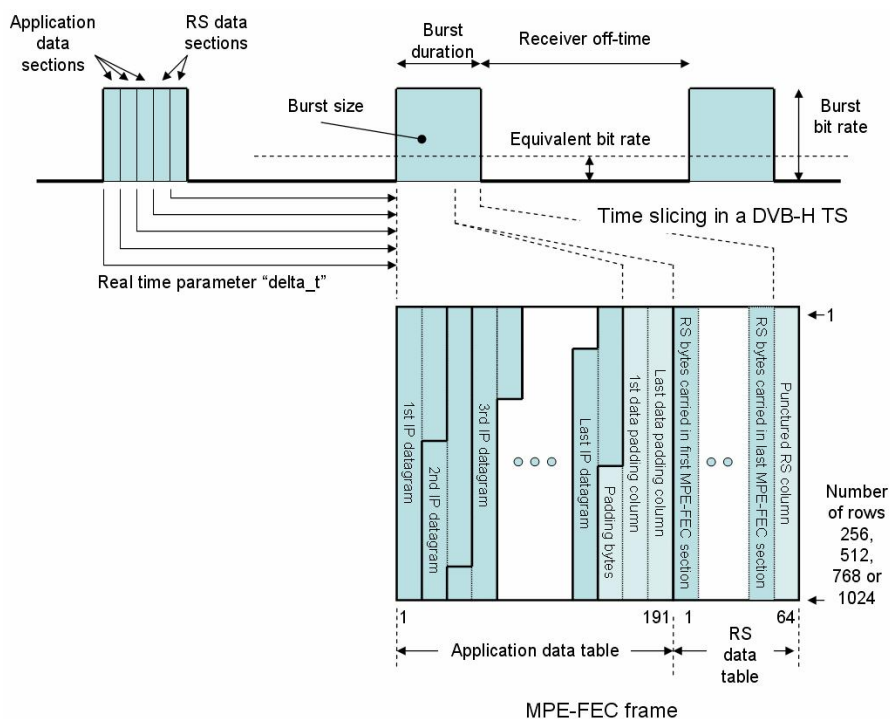
Fig. 18: MPE-FEC Frame

4.3 Testing DVB-H Functionality

The functional units specified above should be tested when the receiver is tested. Since the standard allows for a variety of possible receiver-end configurations especially for FEC and time slicing, a more in-depth test requires that the corresponding parameters also be varied in the test signal.

4.4 Test Setup

Fig. 19 shows the test setup for a DVB-H-compatible receiver using Rohde&Schwarz instruments. The transport stream is generated with one of the following approaches:

- R&S DVRG
- Directly in the R&S SFU using option R&S SFU-K20
- As an alternative to the R&S DVRG, the R&S DVM400 with option R&S DVM400-B2 installed can be used. (The R&S DVM400 offers the advantage that it can also be equipped for TS/DVB-H analysis.)

Transport streams can be combined with DVB-H programs by adding the Stream Library R&S DV-DVBH option to the specified instruments/options.

The DVB-H parameters of the transport stream are varied by selecting corresponding transport stream files on the generator. The transport streams can be any of the following:

- Recorded transport streams
- Transport streams of the Stream Library R&S DV-DVBH option
- Transport streams generating by means of the Advanced Stream Combiner R&S DV-ASC option
  (The transport streams can be generated directly on the instrument.)

DVB-H stream library (DV-DVBH)
or streams made using Advanced
Stream Combiner (DV-ASC)

R&S SFU with
TS generator
or external TS
source like:

R&S DVM400 with
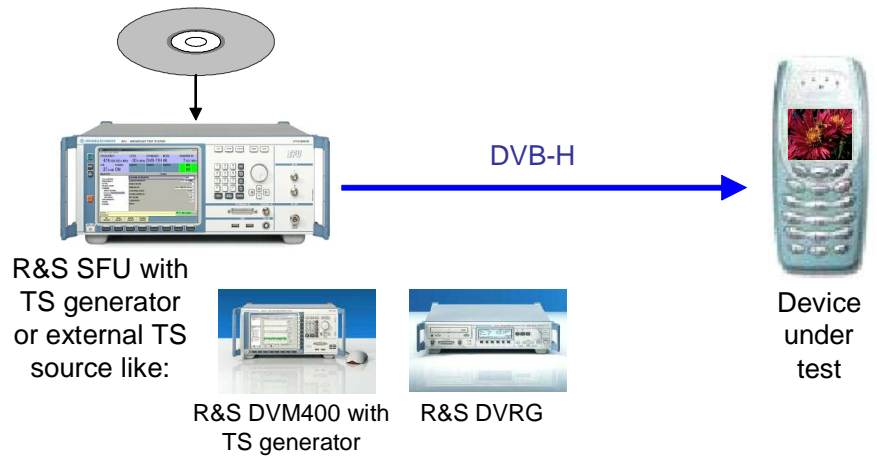TS generator

R&S DVRG

DVB-H

Device
under
test

Fig. 19: DVB-H test scenario

4.5 Tests using the Transport Streams of the DVB-H Stream Library Option

All transport streams of the Stream Library R&S DV-DVBH option include both a DVB-H service and a conventional program with a video and audio elementary stream. They contain all signaling information.

In all transport streams, the MPE-FEC has been calculated for the DVB-H program. In addition, all transport streams are generated in such a manner that the time slicing information is also error-free at the end of the file during replay. This means that no breaks occur at the transport stream level even in the case of endless generation.

Special transport streams are available for testing the following characteristics:

- Maximum data rate performance
- Processing of minimum receiver off-time
- Power consumption as a function of burst cycle time
- Various formats for the useful data:
    - IP version: IPv4 and IPv6
    - Coding standard: MPEG-4, H264 and WM9
    - Resolution: CIF and QCIF

The primary advantage of the Stream Library R&S DV-DVBH option is that it is easy but effective to use. Standard-conforming transport streams for testing DVB-H-compatible receivers can be accessed at the press of a button. Different signals can be selected in order to test all DVB-H-specific functions and special characteristics of the receiver.

4.6 Expanded Test Functions via the Advanced Stream Combiner Option[2]

The Stream Library R&S DV-DVBH option supports a number of different tests. However, cases may arise that require transport streams with special characteristics not found in the library. The following requirements can be covered with additional transport streams:

- Violations of standards
- Special useful data (for example, special video sequences)
- Variation of the MPE-FEC parameters
- Variation of the time slicing parameters
- Additional combinations of the DVB-H parameters

The Advanced Stream Combiner R&S DV-ASC option makes it easy to generate transport streams. Users can vary all specified characteristics as needed, including those of the time slicing and MPE-FEC frame parameters. If the Advanced Stream Combiner option is installed directly on the R&S DVRG, R&S DVM400 or R&S SFU, the generated transport streams can be used directly without having to perform any additional steps. Video signals converted to IP data are available as useful data. The software tools supplied with the Advanced Stream Combiner option make it

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[2] Planned availability: April 2005

easy to convert even video signals to IP data for insertion into a DVB-H program.

5 Broadcast Test System R&S SFU – RF Signal Source for DVB-H

The Broadcast Test System R&S SFU is an excellent platform for numerous digital TV applications: R&D, production, quality control, service, propagation and reception tests by network operators, and EMC measurements.

The R&S SFU occupies only four height units and offers unique RF and baseband features for a wide variety of applications. Its modular design makes it easily adaptable to the requirements at hand.

Fig. 20: Broadcast Test System R&S SFU – ideal for tests with DVB-H

The R&S SFU is ideal for R&D when existing standards need to be improved and new standard signals generated. It covers applications that previously required many different instruments.

Due to its modern and intuitive concept, the R&S SFU is fast and easy to operate.

Switching between operating parameters (e.g. roll-off, puncture rate, QAM mode) is easy. Values exceeding those defined in the standard are possible for lab applications. For special tasks such as in DVB-T/H, the modulation, individual carriers and carrier groups can be deactivated. Sweeps of the entire RF range are possible.

The instrument's high signal quality permits exact measurements, made possible by digital signal processing in the baseband.

The optionally integrated transport stream generator supplies digital test signals in the baseband for the I/Q modulator. These signals can also be made available at the ASI output for external instruments.

The various predefined transport streams for the generator can be selected from the hard disk, where they are stored in .GTS file format. The transport stream generator allows high-bit-rate MPEG-2 transport streams for the broadcast range to be generated endlessly and seamlessly.

As an expansion, additional transport stream libraries, including for DVB-H, can be integrated into the transport stream generator. The R&S DV-DVBH option expands test functionality to cover the new mobile broadcasting standard DVB-H (see "Tests using the Transport Streams of the DVB-H Stream Library Option"). Fast, reproducible tests for mobile receivers such as PDAs and mobile phones can be provided by means of the signals.
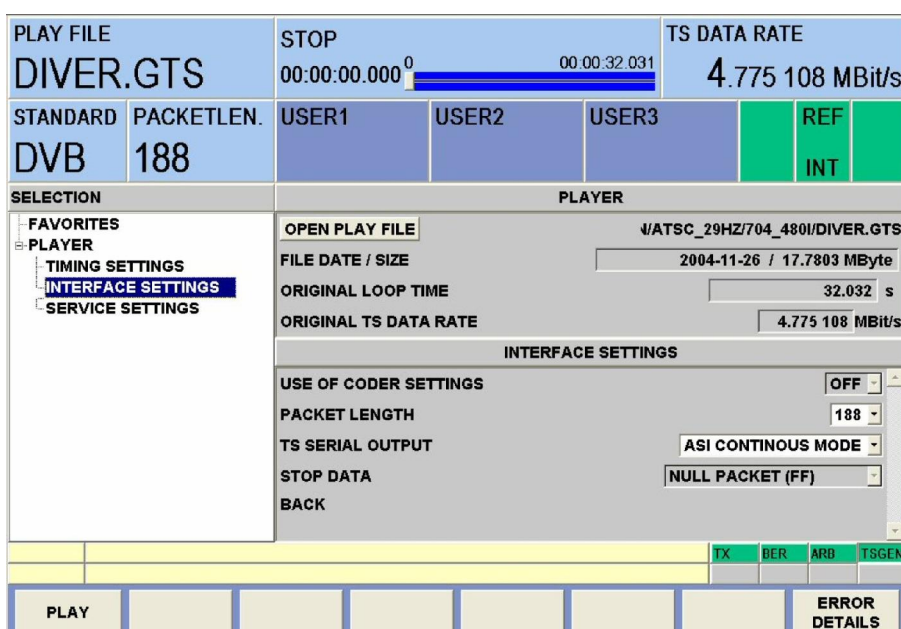

Fig. 21: Windows interface of the transport stream generator application

To enable users to record their own transport streams, a transport stream recorder and player is available as an expansion. Any number of external transport streams with data rates between 100 kbit/s and 216 Mbit/s (including null packets) can be recorded either via the serial interface (ASI) or the parallel interface (SPI) by using this expansion.

Only the size of the hard disk limits the amount of data that can be recorded.

Once recorded, transport streams can be replayed endlessly with packet-exact cutting at the end-of-file/start-of-file transition.

The DVB-T/H coder provides the R&S SFU with the characteristics standardized for DVB-T and DVB-H.
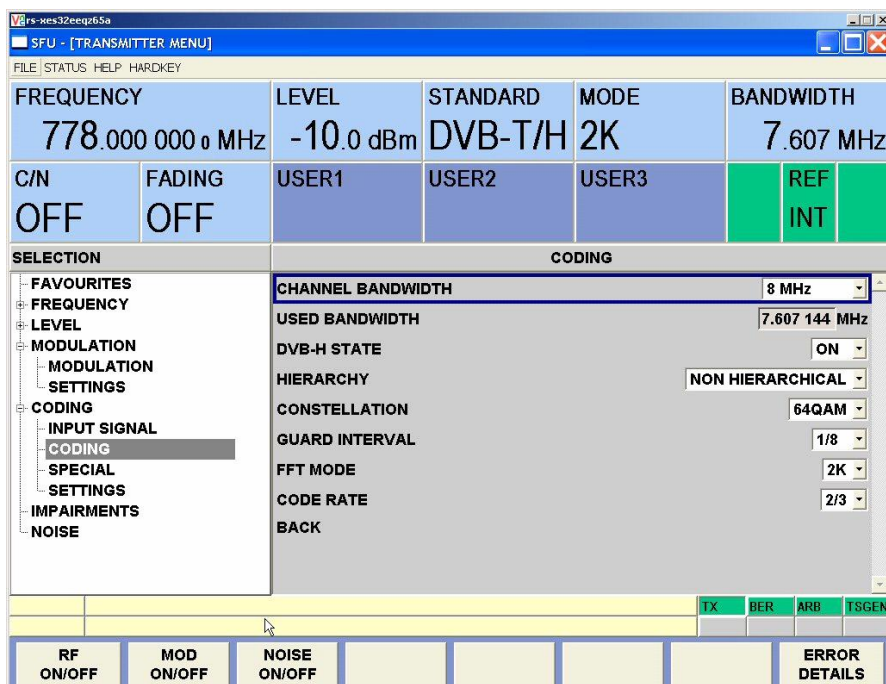
Fig. 22: Settings for the internal test signal

Of course, the new DVB-T/H coder includes the 6 MHz, 7 MHz and 8 MHz bandwidths already in use with DVB-T. But it also offers the 5 MHz bandwidth adopted with the DVB-H standard.

In the case of DVB-T/H coding, the R&S SFU allows the following to be selected: the source, an internal transport stream generator (.GTS file format if a predefined transport stream from a library is to be used) or a transport stream recoder (.TRP file format if a transport stream has been recorded and is to be replayed), or an external signal.

An external transport stream can be fed to the R&S SFU in parallel form via the SPI interface or in serial form via the ASI interface. Moreover, the R&S SFU also has additional internal test signals.
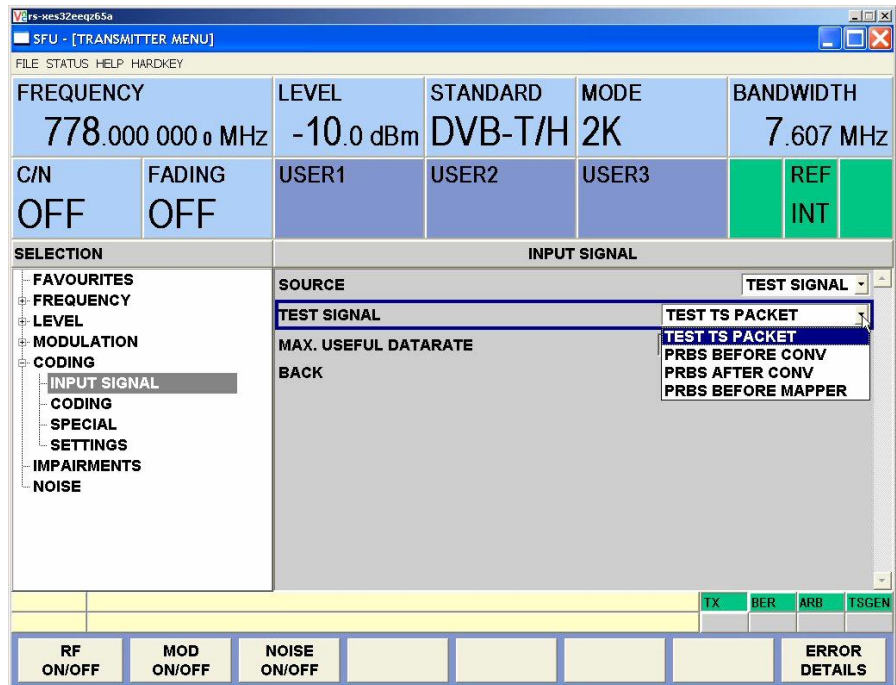
Fig. 23: Settings for the internal test signal

Users can set the type and structure of the supplied transport stream with packet lengths of either 188 bytes or 204 bytes. In addition, either continuous mode or burst mode can be set for the serial interface. The R&S SFU also allows stuffing to be turned on or off.

The maximum useful data rate depends on the settings of the modulation parameters (for example, DVB-T/H). The input data rate of the transport stream is measured accordingly. Thus, users can immediately see the input data rate. The useful data rate is calculated from the transport stream actually supplied to the R&S SFU. Only transport packets with true useful data (user payload) are taken into consideration. Stuffing bytes and null packets are excluded.
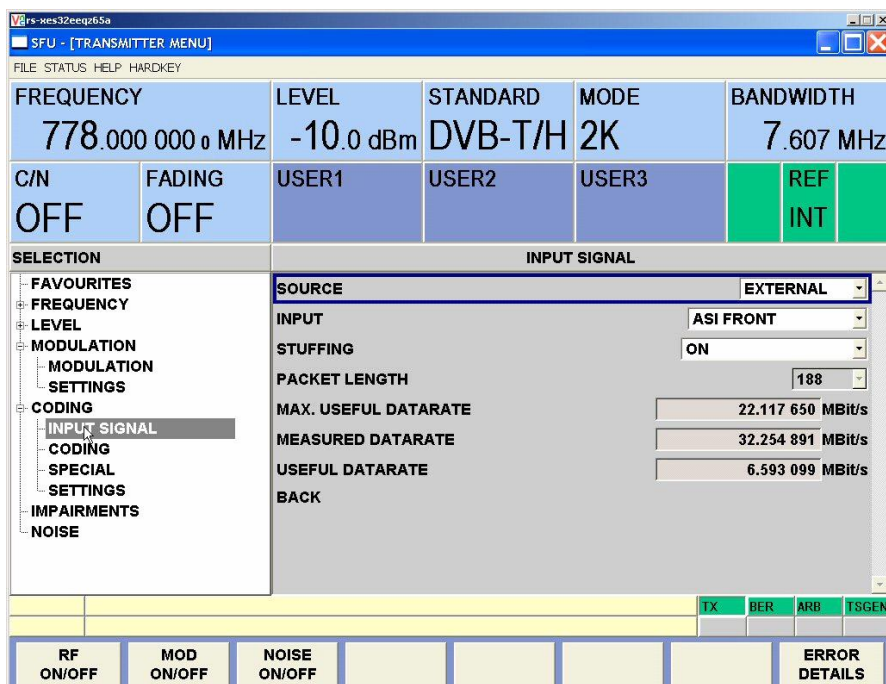
Fig. 24: Settings for the input signal

The R&S SFU offers expanded DVB-H functionality. All DVB-H-relevant parameters can be activated or deactivated.

This includes the DVB-H state. If the DVB-H state has been deactivated, the DVB-T/H coder operates in accordance with EN 300744, i.e. like a pure DVB-T transmitter. When set to native mode, the symbol interleaver is compatible with the existing DVB-T receivers that offer the conventional 2k and 8k modes.

In the FFT mode, it is possible to switch between 2k, 4k and 8k mode for DVB-H. The 4k mode characteristics are specially optimized for mobile transmission. This also affects the duration of the guard interval.

If the DVB-H state is activated, the symbol interleaver allows either the native or in-depth mode to be selected.

Owing to its large memory depth (and time-extended/spread signal), the in-depth mode is more robust against interferences that may occur in a real mobile reception environment in which conditions are constantly changing. Interferences also include impulsive noise.

The TPS bits are used to activate signaling for transport streams with time slicing and to activate MPE-FEC. TPS bit signaling can be set to either S48 or S49.

During transmission, MPE-FEC (error protection on the IP end prior to the IP in MPEG-2 encapsulation) provides additional error protection while the receiver is evaluating the time slicing signaling. Time slicing in the transport stream helps save power in handheld reception, because the mobile terminal expects the ready-to-receive state for only a brief preannounced time.

Cell IDs can also be changed on the R&S SFU. One use of cell IDs in multifrequency networks (MFN) is to identify transmitters.
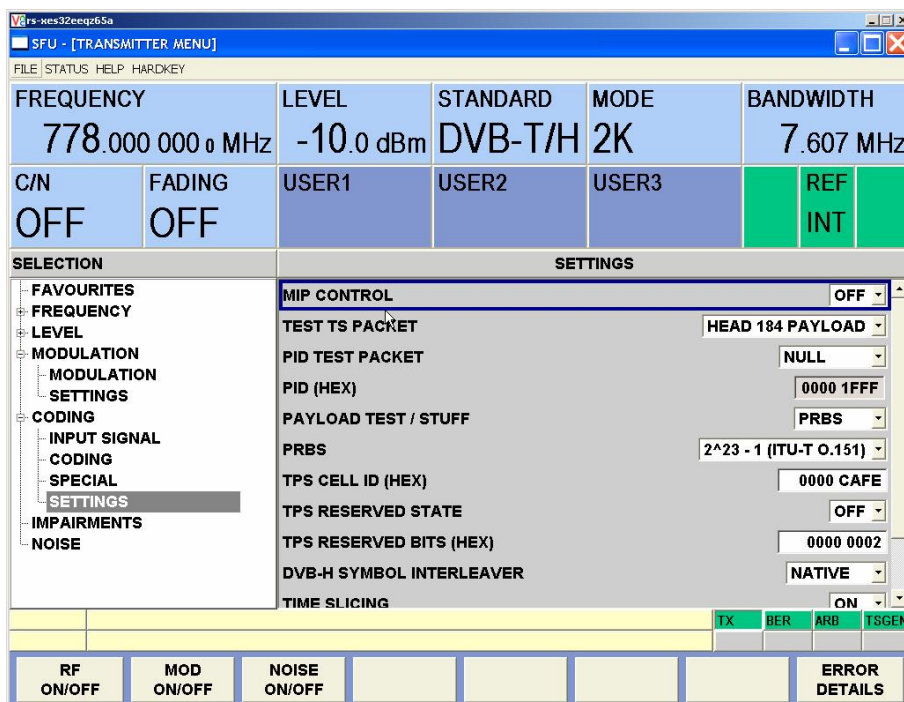
Fig. 25: DVB-H-specific settings for TPS, interleaver, time slicing

In DVB-T/H hierarchical coding, transmission is divided into a high priority and a low priority path. This requires two independent signals: one for the high priority path and one for the low priority path. For both paths, this also means that it must be possible to define all settings independently of each other, e.g. signal source, packet length, stuffing and data rate.

Two methods of implementation are possible. One coding method utilizes only one external transport stream and the additionally internally generated transport stream (PRB sequence, or transport stream generator). Another method is to use two different external transport streams. Both transport streams must then be fed to the R&S SFU from an external source.

Of course, other signal combinations are also possible; the following table lists several.

Source HP	Source LP
External	External
Test signal	External
External	Test signal
Test signal	Test signal
TS generator	External
TS generator	Test signal
External	TS generator
Test signal	TS generator

Table 2: Possible signal source combinations in hierarchical coding

The high priority signaling for time slicing and for MPE-FEC is transmitted in frames 1 and 3, while the low priority signaling is transmitted in frames 2 and 4.

All standard DVB-T functions are also available to R&S SFU users.

DVB-H technology allows video streaming to be implemented efficiently for portable and mobile reception with small devices and mobile phones.

6   Test Receiver R&S EFA for RF Measurements of the DVB-H Signal

As mentioned earlier, DVB-H is an expansion of the DVB-T standard. Therefore, a number of measurements can also be performed on a DVB-H signal by using a conventional DVB-T receiver. Models 40 and 43 of the R&S EFA family of test receivers from Rohde & Schwarz have already shown themselves to be well-suited for DVB-T applications in development, production and broadcasting. Moreover, analog models of the R&S EFA family can also be combined with the R&S EFA-B10 option for DVB-T.

To support DVB-H, the latest firmware version (5.20) has been added to the DVB-T models of the R&S EFA family so that the new DVB-H-specific TPS information can be detected.

Fig. 26 shows the new R&S EFA measurement screen. Table 3 explains the DVB-H-compliant interpretation of the reserved TPS bits. The state interleaver state (S27, native or in-depth) and the TPS length indicator are also specified.

Fig. 26: R&S EFA measurement screen (model 40 or 43 or models with R&S EFA-B10)

Position	7 (MSB)	6	5	4	3	2	1	0 (LSB)
TPS	0	0	S48	S49	S50	S51	S52	S53

S48	S49	DVB-H signaling
0	x	Time slicing not used
1	x	At least on elementary stream uses time slicing
x	0	MPE-FEC not used
x	1	At least on elementary stream uses MPE-FEC
Note: "x" indicates either bit state		

Table 3: New TPS bits for DVB-H

The following example makes this clear:

Information on the display:

"**TPS RES (F1-F4): 30, 30, 30, 30"** (Hex)

$\Rightarrow$ Binary: **0011** 0000, 0011 0000, 0011 0000, 0011 0000

In this example, TPS bits S48 and S49 are thus set to 1. All other TPS bits in this frame (F1) are set to 0. According to table 3, this means that time slicing and MPE-FEC are active.

The screen shown in Fig. 26 provides a quick overview of the main parameters – e.g. frequency offset, MER, BER and modulation modes – that are displayed by the Test Receiver R&S EFA in the 2k and 8k modes when native interleaving is active. To allow further analysis of the DVB-H signal, additional screens without any restrictions are available in the 2k and 8k modes when native interleaving is active. These screens include the constellation diagram, spectrum, complementary cumulative distribution function (CCDF), impulse response (Fig. 27), and the MER across all OFDM carriers (Fig. 28), which is useful for detecting interference.
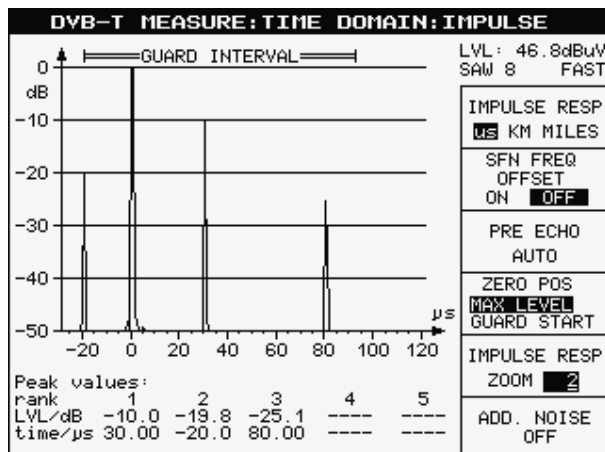
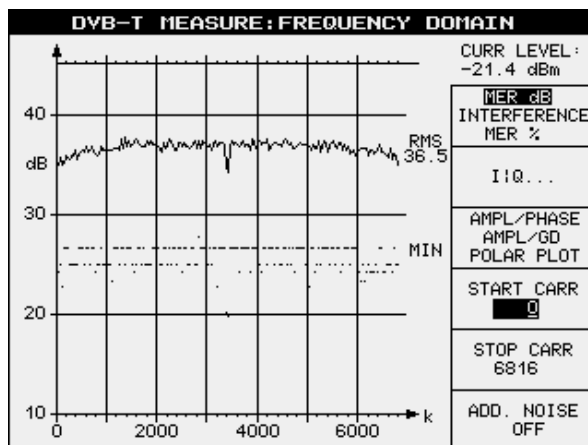Fig. 27: Impulse response with display of guard interval

Fig. 28: MER across all OFDM carriers

When the new DVB-H in-depth interleaving is used in the 2k mode, all measurements specified above can be carried out at the modulation layer by using the R&S EFA if FEC SYNC NOT REQUIRED has been selected under SPECIAL FUNCTIONN:SYNC CONDITION. Depending on the system involved, the only measurement that may not be possible here is BER measurement.

The new 4k mode enables the R&S EFA to display the level, spectrum and CCDF. For example, the quality of a DVB-T/H modulator is generally determined by performing measurements in the 2k or 8k mode. These measurements provide sufficient information about the modulation characteristics in the 4k mode.

6.1 Single Frequency Networks (SFN)

Particular attention must be paid to the TPS length indicator when putting SFNs into operation. Reliable and error-free operation requires that all

components of the OFDM signal of various transmitters be identical. This also means that TPS information must be correct. This information must be identical in all cases in order to prevent signal decoding errors. The R&S EFA is the ideal tool for performing thorough measurements of all transmitted TPS bits.

Table 4 provides a complete overview of all DVB-H measurements that can be performed using the R&S EFA.

The DVB-H characteristics of the MPEG-2 transport stream demodulated by the R&S EFA can be analyzed by means of the MPEG-2 Monitoring System R&S DVM50/100/400, which provides information about the MPE and FEC sections. Time slicing can also be measured.

	2k mode, native inter-leaving	2k mode, in-depth inter-leaving *)	4k mode	8k mode
Level	yes	yes	yes	yes
Constellation	yes	yes	--	yes
OFDM parameters	yes	yes	--	yes
MER/EVM	yes	yes	--	yes
MER(f)	yes	yes	--	yes
I/Q(f)	yes	yes	--	yes
Impulse response	yes	yes	--	yes
BER before Viterbi	yes	--	--	yes
BER before RS	yes	--	--	yes
BER after RS	yes	--	--	yes
Spectrum/CCDF	yes	yes	yes	yes
DVB-H signaling (TPS)	yes	yes	--	yes
TPS length indicator	yes	yes	--	yes

*) "FEC sync not required" must be set

Table 4: DVB-H measurements with R&S EFA model 4x option EFA-B10

7   Rohde&Schwarz DVB-H Broadcasting Equipment

In addition to the DVB-T/H T&M equipment described in detail above, Rohde&Schwarz also offers low-, medium- and high-power DVB-T/H transmitters.

DVB-T transmitters of the R&S SX/NX7000 series can be updated to DVB-H. The DTV IP Inserter and Generator R&S DIP010 makes it possible to insert IP packets into MPEG-2 transport streams in compliance with DVB-H by using MPE.

Fig. 29: ATV/DTV transmitters from Rohde&Schwarz

Fig. 30: DTV IP Inserter and Generator R&S DIP010

8 Literature

ETS 300 744: DVB-T Digital Terrestrial Video Broadcasting

ETS 300 468: DVB Service Information

ETS 301 192: DVB Specification for Data Broadcasting

ISO/IEC 13818 MPEG-2 Standard

ISO/IEC 14496 MPEG-4 Standard

Walter Fischer, Digital Televison – A Practical Guide for Engineers, Springer, ISBN 3-540-01155-2

9 Additional Information

This application note is updated from time to time. Please visit the website **1MA91** in order to download new versions.

Please contact TM-Applications@rsd.rohde-schwarz.com for comments and further suggestions.

10 Ordering information

Digital Video Measurement System
DVM400   2085.1800.02

MPEG-2 Monitoring System
DVM100   2085.1600.02

Digital Video Recorder & Generator
DVRG   2083.1302.02

Broadcast Test System
SFU   2110.2500.02

TV Test Receiver
EFA   4.5…1000 MHz   2067.3004.43

DTV IP Inserter and Generator
DIP010   3540.9254.00

Universal Radio Communication Tester
CMU200   1100.0008.02

Protocol Tester for WCDMA/HSDPA
CRTU-W   1140.0509.02

Protocol Tester for GSM
CRTU-G   1140.0009.02

ROHDE & SCHWARZ GmbH & Co. KG · Mühldorfstraße 15 · D-81671 München · P.O.B 80 14 69 · D-81614 München ·
Telephone +49 89 4129 -0 · Fax +49 89 4129 - 13777 · Internet: http://www.rohde-schwarz.com