DIGITAL TERRESTRIAL TELEVISION BROADCASTING

by PAUL G.M. DE BOT\textsuperscript{1} and FLAVIO DAFFARA\textsuperscript{2}

\textsuperscript{1}Philips Consumer Electronics, Digital Video Communication Systems, PO Box 80002, 5600 JB Eindhoven, The Netherlands
\textsuperscript{2}Laboratoires d'Electronique Philips S.A., 22 Avenue Descartes, 94453 Limeil-Brévannes, France

Abstract

In the coming years, the current analog television distribution will be replaced by digital distribution. Standards for digital transmission via satellite and cable have been developed for this purpose, and a standard for digital terrestrial is on its way. In this paper, the technical details of digital terrestrial television broadcasting will be described.

Keywords: broadcasting, DVB, terrestrial transmission, television, OFDM.

1. Introduction

Over the past few years, practical systems for video source coding have been developed in the framework of the Moving Pictures Expert Group of the International Standardization Organization (ISO/MPEG). This effort has led to a growing interest for introduction in Europe of digital broadcasting services in the near future. We should distinguish three different means of distribution: satellite direct-to-home distribution, cable network distribution and terrestrial distribution. Since these distribution media each have different channel characteristics and require different receiver equipment, different transmission mechanisms have to be designed, each optimized for a specific medium. All these mechanisms enable the transport of 24 to 40 Mbit/s in a single channel. Since MPEG-2 source coding can provide good standard definition video quality at bit rates of 4 to 8 Mbit/s, such a transport stream is sufficiently large to contain a number (4 to 8) of normal standard definition TV programs. In Fig. 1, the migration from analog transmission to digital transmission is depicted for the different television transmission media. A generic picture of a digital television chain is given in Fig. 2.
A draft European standard describing a transmission mechanism for TV broadcasting via satellite [1] was fixed at the beginning of 1994. Satellite transmission is characterized by low available transmitter power, relatively high channel bandwidth (33 to 40 MHz), highly nonlinear transmitter amplification and a transmission medium which approaches the Additive White Gaussian Noise (AWGN) channel. For these reasons, Quaternary Phase Shift Keying (QPSK) modulation was chosen in combination with a powerful concatenated error correction coding.
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For cable TV networks, another transmission standard was drafted in 1994 [2]. The cable channel is characterized by a high signal-to-noise ratio, a severe bandwidth limitation (8 MHz), and short signal reflections due to impedance mismatches in the network. These constraints have led to the choice of Quadrature Amplitude Modulation with 64 signal points (64-QAM) and interleaving in combination with a single Reed–Solomon code. For compatibility reasons, the interleaving and Reed–Solomon coding are the same as for the satellite system.

The terrestrial channel is the worst and the most difficult of the three channels discussed. This is due to the fact that the Digital Terrestrial Television Broadcasting (DTTB) system should allow large coverage for fixed receivers (with a directional roof-top antenna), and also provide the largest possible coverage for portable receivers (indoor reception with a non-directional built-in antenna). For this reason, no final standard has yet been fixed in Europe. Discussions in all three European DTTB projects\(^1\) focus on the use of Orthogonal Frequency Division Multiplexing (OFDM), in contrast to the single carrier systems chosen for satellite and cable. Also in Japan, OFDM-based systems are considered for DTTB, although in North America a Single Carrier Vestigial Side Band (VSB) modulation solution seems to be preferred.

In the rest of this paper, an OFDM-based DTTB system will be described in more detail.

2. Channel characteristics for terrestrial broadcasting

As mentioned in the introduction, DTTB should allow two different reception conditions, which are related to two different transmission channels. Fixed reception coverage will be mostly interference limited, where the interferer, in the DTTB introduction period, probably would be a PAL/SECAM signal. The transmission channel for portable reception, however, will mainly be characterized by multipath propagation, resulting in a frequency selective, noise limited channel. Single Frequency Networks (SFNs), as will be described in Section 9, produce an effect similar to multipath propagation, also for fixed receivers.

The conventional network planning is based on fixed reception. In Ref. [6], it is shown that transmitters should increase their power by about 30 dB

\(^1\) The RACE dTTb project [3], the German HDTV project [4] and the Nordic Divine project [5].
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to offer the same service area for portable receivers. Therefore, in their introduction phase, the DTTB services will focus on fixed reception.

On noisy multipath channels, the received signal $r$ can be modelled as

$$r = a_M(f)s + n,$$

where $a_M(f)$ is the complex attenuation factor of the channel, $s$ is the transmitted signal with $E[ss^*] = E_s$, and $n$ is a complex AWGN component with $E[nn^*] = N_0$. Usually, $a_M(f)$ is complex Gaussian distributed, and frequency-dependent. Now we can define the frequency-dependent signal-to-noise ratio

$$\gamma_N(f) = |a_M(f)s/n|^2 = |a_M(f)|^2E_s/N_0.$$  

(2)

A typical profile for $\gamma_N(f)$ is shown in Fig. 3.

If a channel suffers from co-channel interference (CCI) that has a complex Gaussian amplitude distribution, the received signal can similarly be written as

$$r = s + \beta(f)n,$$

where $\beta(f)n$ represents the combined CCI and AWGN. If the CCI is caused by PAL/SECAM signals, $\beta$ will be heavily frequency-dependent and show power concentrations near the luminance, chrominance and sound carriers of the PAL/SECAM signal.

As well as for the case of multipath propagation, we can define the frequency-dependent signal-to-noise ratio (where in this case the ‘noise’ is in fact interference) as

$$\gamma_I(f) = |s/\beta(f)n|^2 = |\alpha_I(f)|^2E_s/N_0,$$

(4)

where $\alpha_I(f) = 1/\beta(f)$. A typical $\gamma_I(f)$ profile, for a channel suffering from CCI of PAL, is shown in Fig. 3.

Hence, we can deal with CCI from PAL/SECAM in the same way as with multipath propagation. Channels suffering from CCI from PAL/SECAM as well as channels with multipath propagation will be referred to as frequency selective channels. In the following, we will use the term signal-to-noise

![Fig. 3. $\gamma_N(f)$ profile of multipath reception in an urban area (left) and $\gamma_I(f)$ profile of PAL co-channel interference (right). The wideband value of $\gamma$ equals 0 dB in both cases.](image)
ratio (SNR), although we mean the signal-to-noise-plus-interference ratio \( S/(I + N) \).

3. Orthogonal frequency division multiplexing

In Section 2, multipath transmission has been discussed, which is a predominant factor limiting the service coverage for terrestrial broadcasting.

Let us assume to have a channel allowing the transmission of symbols with duration \( T_s \). If the channel introduces echoes, their effect is limited if the maximum echo delay \( \tau \) is small compared to \( T_s \). To improve the resistance to echoes further, each symbol of duration \( T_s \) can be extended with a so-called guard interval of length \( T_g \), containing a cyclic continuation of the symbol. This leads to symbols of total duration \( T_s + T_g \) but reduces the transmission efficiency of the channel. If an echo occurs with delay \( \tau < T_g \), the received symbol overlaps with both the previous symbol and the next one. However, a window of width \( T_s \), which is not corrupted by intersymbol interference (ISI), can be found in the centre of the period \( T_s + T_g \). If a receiver is able to properly position observation windows of length \( T_s \) over the received signal, the transmitted symbols can be recovered without suffering from ISI.

However, in SFNs (see Section 9), the echo delay can be as large as 200 \( \mu s \). This means that the guard interval should have a duration of \( T_g = 200 \mu s \). To ensure a sufficiently large transmission efficiency, the (Nyquist) symbol period should be chosen not smaller than \( T_s = 800 \mu s \), yielding an efficiency loss of 20\% due to the guard interval insertion. If we transmit the symbols with a rectangular pulse shape in the time domain, the Fourier transform of the signal \( s(t) \) will be

\[
S(f) = T_s \text{sinc}( (f - f_c) T_s )
\]

where \( f_c \) is the frequency of the carrier. If \( T_s = 1 \text{ ms} \), the effective bandwidth \( F_s \) of the signal is 1 kHz. Since channels of 8 MHz are available for DTTB, we could combine many such narrowband signals in the wideband transmission channel. If we use signals \( s_k(t) \) with carrier frequencies \( f_{c,k} \), each exactly \( F_s = 1 \text{ kHz} \) apart, the signals are orthogonal. This means that at the receiver side the different signals \( s_k(t) \) can be recovered without any mutual cross-talk. This technique is known as Orthogonal Frequency Division Multiplexing (OFDM). OFDM is proposed for DTTB transmission in Europe and Japan. OFDM is also being used in the Digital Audio Broadcasting (DAB) system [7,8].

To combine the large number of narrowband signals into a wideband OFDM signal, an Inverse Discrete Fourier Transform (IDFT) can be used.
at the transmitter side, combined with a Discrete Fourier Transform (DFT) at the receiver side. By using a complex IDFT of $N = 8192$ points, we can multiplex $N$ signals $s_k(t)$ with $k = 0, \ldots, N-1$ onto an 8 MHz channel. In Europe, this so-called 8 K OFDM scheme is currently proposed for DTTB.

Since guard bands in the frequency domain are needed for filtering, a number of carriers at the edges of the 8 MHz channel are left unmodulated (virtual carriers). Only about $N_{\text{eff}} = 7500$ carriers are modulated, giving an effective bandwidth of some 7.5 MHz.

Hence, in each OFDM time slot (with a duration of $T_s + T_g$) we can transmit $N_{\text{eff}}$ complex symbols. These symbols can be Phase Shift Keying (PSK) or Quadrature Amplitude Modulation (QAM) symbols, as will be described in Section 4. The total gross symbol rate over a channel equals $N_{\text{eff}}/(T_s + T_g)$. An example of an OFDM scheme is given in Table I.

By using OFDM with guard intervals, the problem of ISI in the time domain is solved. However, the frequency selective nature of the channel (due to both multipath and CCI) causes each of the OFDM carriers $s_k$ to be subject to a different signal-to-noise ratio $\gamma_k = \gamma(f_{c,k})$. Error correction coding is needed to recover the information transmitted on the carriers which are subject to low $\gamma$ values. Error correction coding will be described further in Section 5. In addition to error correction, the frequency selectivity can be reduced by using antenna diversity with narrowband combining. This technique can improve the performances on severe frequency selective channels by up to 10 dB [9,10].

The output of the IDFT at the transmitter side is a signal of which the amplitude has a very high dynamic range and the peak-to-average power

<table>
<thead>
<tr>
<th>TABLE I</th>
<th>Example of an OFDM scheme for nationwide SFNs</th>
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</thead>
<tbody>
<tr>
<td>DFT size</td>
<td>$N$</td>
</tr>
<tr>
<td>Sampling frequency</td>
<td>$B$</td>
</tr>
<tr>
<td>Symbol period</td>
<td>$T_s = N/B$</td>
</tr>
<tr>
<td>Carrier spacing</td>
<td>$F_s = 1/T_s$</td>
</tr>
<tr>
<td>Guard interval</td>
<td>$T_g = T_s/4$</td>
</tr>
<tr>
<td>Effective number of subchannels</td>
<td>$N_{\text{eff}}$</td>
</tr>
<tr>
<td>Effective signal bandwidth</td>
<td>$B_{\text{eff}} = N_{\text{eff}}F_s$</td>
</tr>
<tr>
<td>Gross symbol rate</td>
<td>$R_g = N_{\text{eff}}/(T_s + T_g)$</td>
</tr>
<tr>
<td>Net symbol rate</td>
<td>$R_n = R_g(93/96)(7/8)$</td>
</tr>
</tbody>
</table>
ratio is very large (see Fig. 4). Therefore, clipping will inevitably occur at the high-power transmitter amplifiers, due to their non-linear behaviour. The percentage of signal which is clipped depends on the output back-off of the power amplifier. The larger the percentage of signal which is clipped, the more the performance at the receiver side is degraded. Since broadcasters want to operate with cost-effective power amplifiers, a trade-off has to be determined between the nominal amplifier power (and the used output back-off) and the acceptable degradation due to clipping. Another way to improve the performance at the receiver side is to use predistortion techniques on the signal before transmission [11].

In Fig. 5, the DTTB transmission chain is shown.

4. Modulation

As explained in Section 3, we can modulate one complex symbol in each timeslot on each useful carrier. These symbols are elements of a symbol set. If we put all the symbols of a symbol set in a complex plane, we obtain what is called the signal constellation. Typically, a signal constellation contains $M = 2^m$ signal points, which means that each symbol carries $m$ bits of information. Hence, $M$ should be large to obtain a large transmission rate. On the other hand, if $M$ is large, the required signal-to-noise ratio to obtain a desired error rate is also large. The choice of a signal constellation is made according to a trade-off between transmission rate and required signal-to-noise ratio.

For DTTB, usually three modulation schemes are considered; 4-PSK (or...
MPEG-II Transport Packets

MPEG-II Transport Packets

Energy Dispersal Scrambling

RS Encoder

Convolutional Encoder (84 states)

Concatenated Encoder

Frequency Bit Interleaver

64-QAM Constellation Mapping

OFDM (IFFT)

Transmitter Front-end

Terrestrial Channel

Demodulator

Energy Dispersal Descrambling

RS Decoder

Convolutional Decoder (84 states)

Concatenated Decoder

Frequency Bit Deinterleaver

64-QAM Constellation Demapping

OFDD (FFT)

Receiver Front-End

Synchronization

Fig. 5. The DTTB transmission chain.
4-QAM), 16-QAM and 64-QAM, with $M = 4$, $M = 16$ and $M = 64$, respectively. To obtain a sufficiently low error rate for these modulation schemes we need signal-to-noise ratios in the order of $E_s/N_0 = 6\,\text{dB}$, $E_s/N_0 = 12\,\text{dB}$ and $E_s/N_0 = 18\,\text{dB}$, respectively. The signal constellations, with added noise of the critical SNR values, are shown in Figs 6 to 8.

The $m$ bits are usually mapped into the constellation points using Gray mapping. In this case, an error event will cause a minimum number of bit errors.

Some DTTB proposals foresee the option of hierarchical transmission. In this case, the modulation and error protection are organized such that at the receiver side, different bit streams can be extracted from the received signal, each with a different a priori reliability. For example, we can use non-uniform QAM signal constellations, transmitting $m_{HP}$ high priority (HP) bits and $m_{LP}$ low priority (LP) bits per symbol. On bad channels, a receiver will only be able to recover the HP bits reliably, while on a good transmission channel, also the LP bits will be detected with a low error probability. Examples of hierarchical DTTB systems are described in [12,13].
Fig. 8. 64-QAM signal constellation with AWGN ($E_s/N_0 = 18$ dB).

5. Error correcting coding

Error correcting coding is used to guarantee at the input of the demultiplexer virtual error-free performances (i.e. a bit error rate (BER) less than $10^{-10}$). To achieve this, the same concatenated error correcting code as in the digital satellite television standard is used. This concatenated code consists of a $\nu = 6$, 64-state (punctured) convolutional inner code with code rates $R = 1/2$, $2/3$, $3/4$, $5/6$ and $7/8$, a Reed–Solomon code with length 204 and dimension 188 over $GF(2^8)$, and appropriate interleaving.

The inner convolutional code has the advantage of being decodable with the Viterbi algorithm, which is an implementable Maximum Likelihood (ML) decoding algorithm. This makes the code a powerful tool to reduce a BER from $10^{-1}$–$10^{-2}$ at the output of the channel to $10^{-3}$–$10^{-4}$. Since the Viterbi decoder is only able to correct random bit errors, and has no burst error correction capabilities, inner bit interleaving in the frequency domain is applied, to scatter the frequency selective behaviour of the channel and provide random bit errors at the input of the Viterbi decoder. When the Viterbi decoder fails to correct certain errors, it typically produces burst errors of $5\nu$ to $15\nu$ (equivalent to 30 to 90) bits, depending on the puncture rate of the code and of the channel. Since the outer RS-decoder is able to correct random byte errors (but not bursts of byte errors), the bits at the Viterbi decoder output are organized in bytes, on which outer byte interleaving is applied using a Forney interleaver of a depth of 12 bytes. Thanks to its large Hamming distance $d = 17$, which makes correction possible of up to 8 random byte errors per codeword, the Reed–Solomon decoder is able to reduce the bit error rate further, e.g. from $10^{-3}$–$10^{-4}$ down to $10^{-10}$–$10^{-11}$.

The Viterbi algorithm can be implemented for either hard-decision or
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Soft-decision decoding, relative to hard-decision decoding, of a $n = 6$, $R = 1/2$ convolutional code with BPSK modulation can yield a gain of up to 4 dB in the region of $BER = 10^{-4}$ on a Rayleigh fading channel [14]. Because a soft-decision Viterbi decoder uses $r = \alpha s + n$ at its input, the channel state $\alpha$ is implicitly being used by the decoder. Explicit knowledge of $\alpha$, by the availability of Channel State Information (CSI), can yield some additional gain. In the case of frequency selectivity, the optimal method for combining this CSI with the received signal is by applying maximal ratio combining [15]. Using this method, $\alpha^* r$ is supplied to the input of the Viterbi decoder. Hence, strong signals are made stronger, while weak signals are made even weaker. The use of CSI in this way can yield an additional gain of 2 dB in the region of $BER = 10^{-4}$ on a Rayleigh fading channel [14]. It is furthermore interesting to see that if we use hard-decision Viterbi decoding with 1-bit CSI (comparable with an erasure flag), we closely approach the performances of soft-decision Viterbi decoding without CSI [14].

If the channel is primarily subject to interference, rather than to multipath, it is essential that this interference be estimated in order to apply soft-decision decoding.

Methods for estimating the CSI are given in Section 7.

6. Synchronization

Good reception of the transmitted signals is only possible if good frequency and time synchronization is achieved. To enhance synchronization performance, the DFT blocks, or OFDM symbols, are organized in frames. In the European Digital Video Broadcasting (DVB) project, a frame containing a total of 96 OFDM symbols is proposed, including a silent period (null symbol). This null symbol is used for coarse time synchronization. Immediately after the null symbol, a reference symbol having good autocorrelation properties in the frequency domain is transmitted. This reference symbol is used for coarse frequency synchronization. Since the transmitted content of the reference symbols is known a priori at the receiver side, the receiver is able to estimate the frequency domain transfer function $H(f)$. Hence, the receiver can calculate the time domain transfer function $h(t)$, and is consequently able to determine the optimal observation window position (fine time synchronization). The reference symbol in its turn is followed by a Transmission Parameter Signalling (TPS) symbol, which contains transmission mode information, such as the used signal constellation and convolutional code rate. The TPS symbol is modulated and protected such that it can be received even under very bad channel conditions.
Since oscillator stability of the receiver can be a limiting factor, the frequency synchronization needs to be extremely accurate. A small frequency error causes a fixed rate of phase rotation in each QAM signal, or cell, as well as cross-talk between the subcarriers. To support more accurate frequency synchronization, the remaining 93 symbols in a frame contain a certain amount of pilots, which do not contain data and from which the receiver can estimate the actual frequency offset. In order to minimize the cross-talk, this frequency error signal is being fed back and compensated for, prior to the DFT. Since OFDM is very sensitive to frequency jitter and phase noise, the local oscillator in the receiver front-end needs to have a very high frequency accuracy.

In the frame structure described above, 93 of the 96 OFDM symbols in a frame are used for data transmission. However, in these 93 symbols, about 12.5% of the carriers are used to transmit pilot symbols. Hence, the net symbol rate equals about (93/96)(7/8) times the gross symbol rate. For the example of Table I, this yields a symbol rate of 5.14 Mbaud.

7. Channel estimation

Furthermore, channel amplitude and phase estimation are required for enabling coherent detection of the QAM signals, and for providing reliability information to the soft-decision Viterbi decoders. This channel estimation is performed by using the reference symbol and pilot cells. To obtain

![Graph](image)

Fig. 9. Performance of 4-QAM with concatenated coding on a Rayleigh channel, for different inner code rates $R$. 

$E_b/N_0$ (dB)

Bit Error Rate (BER)

$R=1/2$ $R=2/3$ $R=3/4$ $R=5/6$ $R=7/8$

$10^{-1}$ $10^{-2}$ $10^{-3}$ $10^{-4}$ $10^{-5}$ $10^{-6}$ $10^{-7}$ $10^{-8}$ $10^{-9}$ $10^{-10}$

0 2 4 6 8 10 12 14 16 18 20 22 24

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information about the frequency domain characteristics of the CCI the null symbol is analysed. More accurate estimation of the \(S/(I + N)\) ratio in each subcarrier can be obtained by examining the statistics of the received signal [16].

8. System performances

In Figs 9 and 10, the BER curves are given for a few modulation/code combinations on a Rayleigh channel. It can be seen that the required SNR ranges from less than 5 to more than 25 dB, depending on the transmission mode.

Furthermore, we should notice that in case of CCI from PAL (see Fig. 3), the receiver will give sufficiently low error rates for signal-to-interference ratios (SIR) as low as 0 dB (depending on modulation/coding). This remarkable result can be explained by the fact that even with an SIR of 0 dB, most of the OFDM carriers are subject to a narrow-band SIR of more than 20 dB, since the interfering power is concentrated on only a limited number of OFDM carriers. The error correcting codes are easily capable of correcting the errors made in this limited number of interfered OFDM carriers. Its ruggedness towards CCI from analog services is one of the major advantages of OFDM.

The net bit rate can be calculated from the net symbol rate, and depends on the modulation and coding. It is given in Table II.

![Diagram](image)

**Fig. 10.** Performance of 64-QAM with concatenated coding on a Rayleigh channel, for different inner code rates \(R\).
TABLE II
Net bit rate (in Mbit/s) of a typical DTTB system (see Table I), for different modulation schemes and inner convolutional code rates $R$

<table>
<thead>
<tr>
<th></th>
<th>QPSK</th>
<th>16QAM</th>
<th>64QAM</th>
</tr>
</thead>
<tbody>
<tr>
<td>$R = 1/2$</td>
<td>4.70</td>
<td>9.40</td>
<td>14.10</td>
</tr>
<tr>
<td>$R = 2/3$</td>
<td>6.27</td>
<td>12.53</td>
<td>18.80</td>
</tr>
<tr>
<td>$R = 3/4$</td>
<td>7.05</td>
<td>14.10</td>
<td>21.15</td>
</tr>
<tr>
<td>$R = 5/6$</td>
<td>7.83</td>
<td>15.67</td>
<td>23.50</td>
</tr>
<tr>
<td>$R = 7/8$</td>
<td>8.23</td>
<td>16.45</td>
<td>24.68</td>
</tr>
</tbody>
</table>

9. Single frequency networks

A Single Frequency Network (SFN) is a broadcast transmitter network consisting of transmitters with overlapping coverage areas that transmit the same program in the same frequency channel at the same time instant. Consequently, the same signal can arrive at a receiver antenna from different SFN transmitters, each with its own delay, which is related to the distance between receiver and transmitter. The receiver can deal with this effect in the same way as it deals with multipath propagation: the signals arriving from distant transmitters are considered as echoes from the signal arriving from the nearby transmitter.

Since conventional analog transmission schemes (as PAL television) cannot cope with extreme multipath, SFNs were traditionally not possible. However, since OFDM systems with guard intervals are inherently capable of handling multipath, SFNs become practical.

Since SFNs improve the efficiency of spectrum usage considerably, the SFN feature is an important advantage of OFDM systems over analog and single carrier digital systems.\(^2\)

We can distinguish between local SFNs, consisting of a single main transmitter and a few gap-fillers to cover shielded areas, and nationwide SFNs, which consist of a large number of main transmitters.

Distribution of the signal from the central studio to the main transmitters can take place in various ways, using microwave or optical fibre links, satellite feeding, or by mutual in-channel feeding of the transmitters throughout the

\(^2\) Digital single carrier systems would require an extremely long adaptive equalizer at the receiver side, which is very complex compared to an OFDM receiver.
network. Synchronization of the transmitters in an SFN is still an object of intensive study.

In nationwide SFNs, the delay spread can be as large of $T_m = 200 \mu s$, causing the need for a guard interval of at least $T_g = 200 \mu s$ and an 8 K DFT, as described in Sec. 3. As an alternative to the use of a large $T_g$ and DFT size, SFN echoes can be cancelled by mixed time/frequency-domain equalization [17].

10. Service introduction

Many difficulties have to be overcome before DTTB can be introduced in Europe. Receiver ICs still have to be designed, SFN network structures have to be established, including the crucial issue of transmitter synchronization, commercially viable introduction scenarios have to be developed, and channel frequencies have to be allocated.

According to an optimistic scenario, the first introduction of DTTB is planned around 1998. The United Kingdom has already made channel space available for service introduction. It is the intention there to establish a nationwide SFN in UHF channel 35, while throughout the country, in each region, at least two taboo channels are assigned for regional and local-SFN operation. In the London area, it is even expected that seven channels of 8 MHz each can be made available. If on average four TV programmes can be accommodated in each channel, the London area will have access to 28 DTTB programmes.

Other countries which could have some perspectives for service introduction are Denmark, Sweden and The Netherlands. In countries like Germany and Italy, all parts of the broadcast spectrum are completely filled with analog TV services. Due to this lack of available channels, introduction of DTTB in these countries seems to be very difficult.

11. Conclusions

The opportunities for the introduction of DTTB are far from being clear. However, in Europe a technical solution has been developed, albeit at the cost of high receiver complexity. During a period of at least 10 to 15 years after service introduction, simulcast with analog PAL/SECAM will be required, which increases the frequency allocation problem. After switching off the analog services, DTTB with SFNs will improve the spectral efficiency of television broadcasting significantly. This means that on the long term, the actual broadcast spectrum could be made available for non-broadcast applications.
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Authors

Paul G.M. de Bot: Ir. degree (Electrical Engineering), Eindhoven University of Technology, 1989; Eindhoven University of Technology, Department of Information and Communication Theory, 1991; Philips Research Laboratories, Eindhoven, 1991–1995. In his designer's work, he studied coded modulation. At Philips he worked on transmission aspects of digital video broadcasting systems. He was involved in several modules of the RACE dTTb project on digital terrestrial television broadcasting. He is currently working with Philips Digital Videocommunication Systems as product manager, broadcast systems.
Flavio Daffara was born in Monza, Italy, on November 29, 1967. He received the Dipl. Ing. degree in Electronics from Politecnico di Torino, Italy, in October 1991. From October 1990 to September 1991 he attended a one-year Master program in Digital Communications at the Ecole Nationale Supérieure des Télécommunications (ENST), Paris, France. In 1991 he had a training period at the Société Anonyme de Télécommunications (SAT), Paris, where he prepared his Eng. degree thesis working on carrier recovery techniques. In November 1991, he joined the Laboratoires d'Electronique Philips (LEP), Limeil-Brévannes (Paris), France, as an R&D engineer. He was involved in several studies in the digital videocommunications field and took part in the RACE dTTb project on digital terrestrial television broadcasting. His current research interests are in the area of communication theory, modulation techniques, synchronization and adaptive filtering.