INTRODUCTION TO DUOBINARY ENCODING AND DECODING

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With a dozen or so MAC TV signals available from satellites such as TV-SAT2, TDF-1, Olympus, Astra and, shortly, BSB, it is surprising to note that relatively few electronics engineers and satellite-TV reception enthusiasts appear to be aware of essential technical backgrounds to MAC. In line with the theme of the month, communications, this article looks at one aspect of the MAC transmission standard that has received little attention so far: duobinary encoding and decoding of the sound and data block.

Se..ral new TV transmission standards were studied and discussed following the channel and orbital position assignments drawn up by WARC 77. Although these studies resulted in proposals for different systems, all and sundry agreed on the need of analogue picture transmission and digital sound transmission on the basis of a time-multiplex scheme instead of a frequency-multiplex scheme as used up to then for the PAL, SECAM and NTSC systems for existing terrestrial TV broadcasts.

The proposals for A-MAC and B-MAC systems were short-lived because they did not provide a complete separation of the picture and sound blocks at the modulation signal level, and in addition we-- hard to implement in existing satell... TV channel bandwidths (note, however, that B-MAC is still used in Australia) A third standard, C-MAC, developed by the IBA and accepted as well as recommended by the EBU, uses a time-multiplex modulation signal in which the frequency-modulated analogue picture components are interspersed with a 2-4 PSK modulated sound and data block with a data rate of 20.25 Mbit/s. This makes C-MAC suitable for transmission by existing communications service satellites such as the ones in the Intelsat and Eutelsat series, but not for direct-to-home transmission and distribution in TV cable networks.

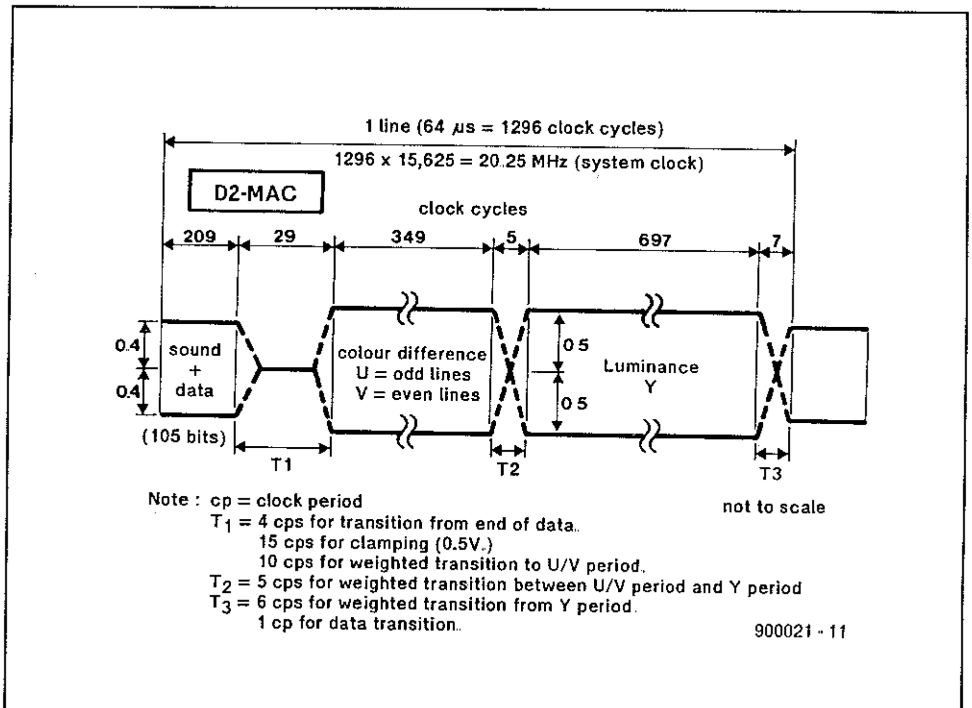

Fig. 1. Time-multiplex division of the picture components and the sound and data block in a D2-MAC signal.

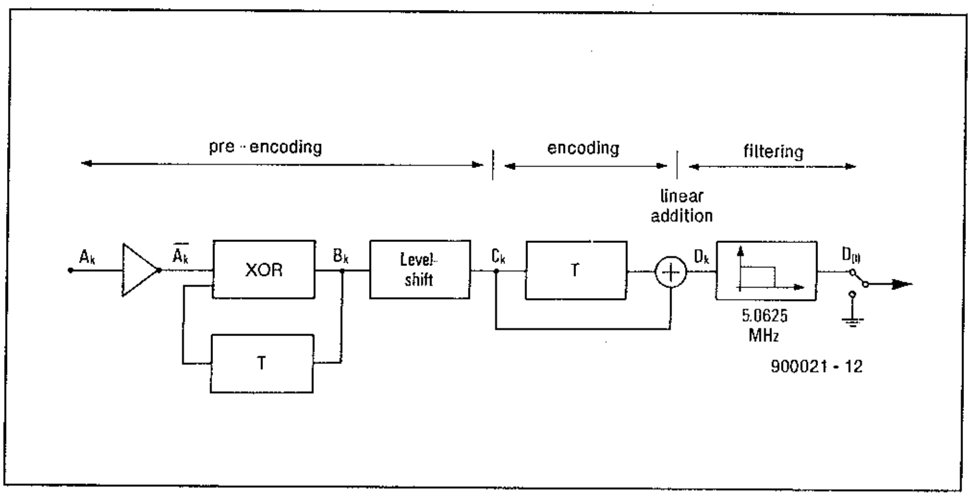

Fig. 2. Basic structure of a duobinary encoder. The two blocks marked 'T' are one-bit time delays (*Illustration courtesy Blaupunkt GmbH*).

D-MAC/Packet and D2-MAC Packet

The need of feeding MAC signals into existing TV cable networks with limited

Sound encoding for D2-MAC/Packet

Sampling rate	1. 32 KHz (HQ) stereo and mono at high quality (40–15,000 Hz) 2. 16 kHz (MQ) Mono at medium quality (40–7,000 Hz)
Quantization	14-bit
Data form	2's complement
Coding	1. Linear, 14-bit (L) 2. NICAM, 10-bit (I)
Error correction	1. Parity check (1) 2. Hamming code (2)
Sound transmission options	1. Parity check for 6 MSB at 10-bit NICAM Capacity: e.g., HQI1 with 4 mono channels 2. Parity check for 11 MSB at 14-bit Linear Capacity: e.g., HQL1 with 3 HiFi mono channels 3. (11,6)Hamming for 6 MSB at 10-bit NICAM Capacity: e.g., HQI2 with 3 HiFi mono channels 4. (16,11)Hamming for 11 MSB at 14-bit Linear Capacity: e.g., HQL2 with 2 HiFi mono channels
Scale factor	covers 32 samples for options 1, 2 and 3 (1 ms at 32 kHz) covers 18 samples for option 4 (562.5 µs at 32 kHz)

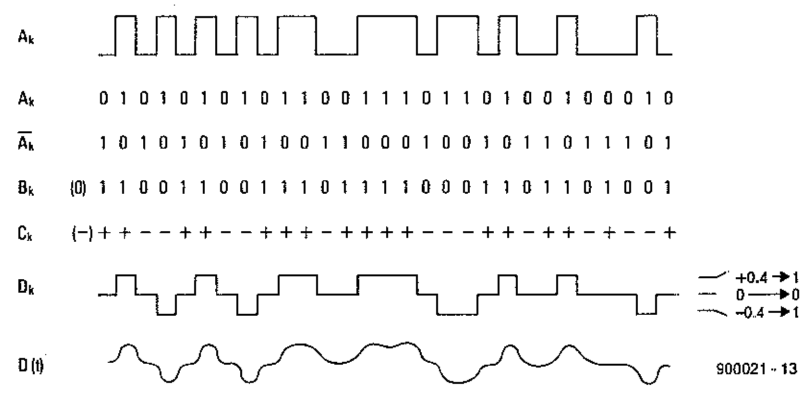

Fig. 3. **Pulse levels and waveforms at various stages of the duobinary encoding process.**

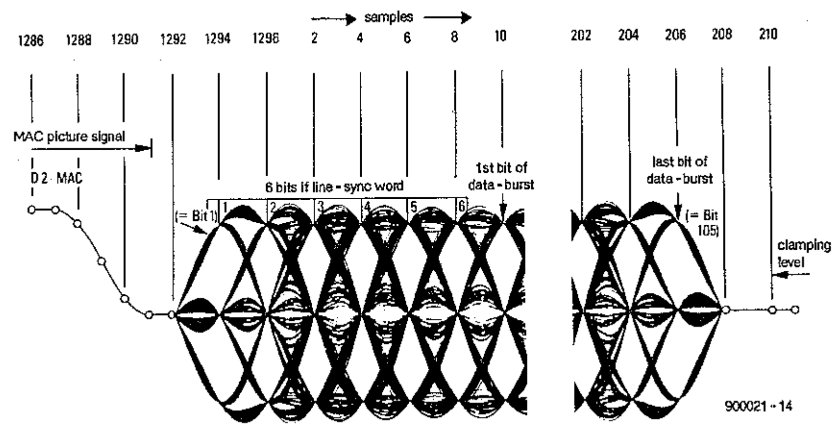

Fig. 4. **Waveform of the three-level signal that forms the data/sound burst in a time-multiplexed MAC picture line. (*illustration courtesy Blaupunkt GmbH*)**

channel bandwidth prompted workers at several radio and television laboratories to look for a means of reducing the bandwidth of C-MAC from about 22 MHz to a value lower than about 10 MHz. The proposals of the CCETT laboratories at Rennes, France, form the basis of the D-MAC/Packet system. This features duobinary encoding of the sound and data block (Packet), so that it can be frequency-modulated like the picture signals, obviating a switch-over at RF level to 2-4 PSK as with C-MAC. The data rate, however, is the same: 20.25 MHz. The D-MAC/Packet system is to be used for all BSB channels to be taken into operation shortly. It offers eight high-quality sound channels, and requires a channel bandwidth of about 10.5 MHz.

The D2-MAC system accepted by most continental European countries is a further development of the D-MAC system. The figure '2' in D2-MAC indicates the bit rate reduction factor with respect to D-MAC. The sampling clock frequency of 20.25 MHz is the same as with C-MAC and D-MAC, but the data bit rate is reduced to 10.125 MHz, so that every second sampling pulse reads one bit. The resultant bandwidth reduction from 10.5 MHz to about 7.8 MHz is sufficient to enable the extensive cable-TV networks on the European continent to carry D2-MAC signals in VSB AM channels. The D2-MAC system supports eight medium-quality or four high-quality sound channels. Both D2-MAC and D-MAC allow NICAM-728 signals to be transmitted. Extensive studies are currently being made into the conversion of both systems into a version that allows HDTV pictures to be transmitted.

Duobinary encoding

The content and digital structure of the sound and data signals are not discussed here, i.e., no consideration will be given to the separation of sound and data, error correction, NICAM-728, compression techniques, quantization factors and the like. The focus of the discussion below is on how the digital data that forms the sound and data to accompany the MAC picture is treated to achieve the bandwidth reduction required for cable-TV systems and direct-to-home reception.

The position of the sound and data block in the time-multiplexed modulation signal fed to the uplink transmitter is shown in Fig. 1. Note that the amplitude is 0.8 Vpp as compared to 1 Vpp for the multiplexed analogue picture components (colour difference and luminance). For D-MAC/Packet, the sound and data block consists of 209 bits instead of 105 as shown for the D2-MAC/Packet system.

The operations involved in duobinary encoding are shown schematically in Fig. 2. The digital datastream A_k is pre-coded to give a datastream Bk. A_k is first inverted, and subsequently combined with the 1-bit delayed (T) result of a XOR operation. In Boolean notation:

$$B_k = A_k \oplus B_{k-1}$$

Next, the 0s and 1s in the datastream B_k are level-shifted to give −1 and +1 levels in the datastream C_k:

$$C_k = 2\,B_k - 1$$

Pre-coding is used to restrain the otherwise unlimited error propagation.

The pre-coded signal is subjected to a delay, T, linearly added to itself, and ,plitude-limited to give datastream D_k.

$$D_k = (M/4)(C_k + C_{k-1})$$

Where M equals 80% of the maximum video amplitude. Amplitude limiting at a fixed factor is required to prevent the duobinary encoded signal exceeding the maximum level of the picture signal.

The datastream D_k can have three instantaneous levels: +0.4 V, −0.4 V and 0 V. The first two represent a logic 1, the last one a logic 0.

The timing diagram in Fig. 2 shows a practical example of how the previously discussed steps convert the digital datastream, A_k, into a duobinary datastream D_k. The reduced bandwidth requirement of D_k is immediately apparent by looking at the duobinary signal that results from the five 0-to-1 transitions at the start of A_k. Subsequent 1s in A_k cause no level change in D_k, which remains at +0.4 V or −0.4 V (remember that the amplitude of D_k in the modulation signal is 0.8 Vpp). For an odd number of 0s in between two 1s, D_k changes from 0.4 V via 0 V to −0.4 V or the other way around. For an even number of 0s, the signal reverts to the previous level.

Referring back to Fig. 1, D_k is passed through a low-pass filter to reduce the bandwidth requirement of the 10.125 Mbit/s (D2-MAC) databurst to about 5 MHz. The addition of the picture components and frequency modulation of the resultant time-multiplexed signal gives rise to a bandwidth of between 7 MHz and 8 MHz (D2-MAC), which is suitable for cable-TV networks.

The filtered three-level component in the modulation signal is illustrated in Fig. 3. Note that the sound and data block is located between the end of the luminance component and the start of the clamping level reference period. The block starts with the 6-bit line synchronization word, LSW, which is either true (LSW) or inverted ($\overline{LSW}$) to indicate whether the line belongs with the odd- or the even-numbered raster in the interlaced picture.

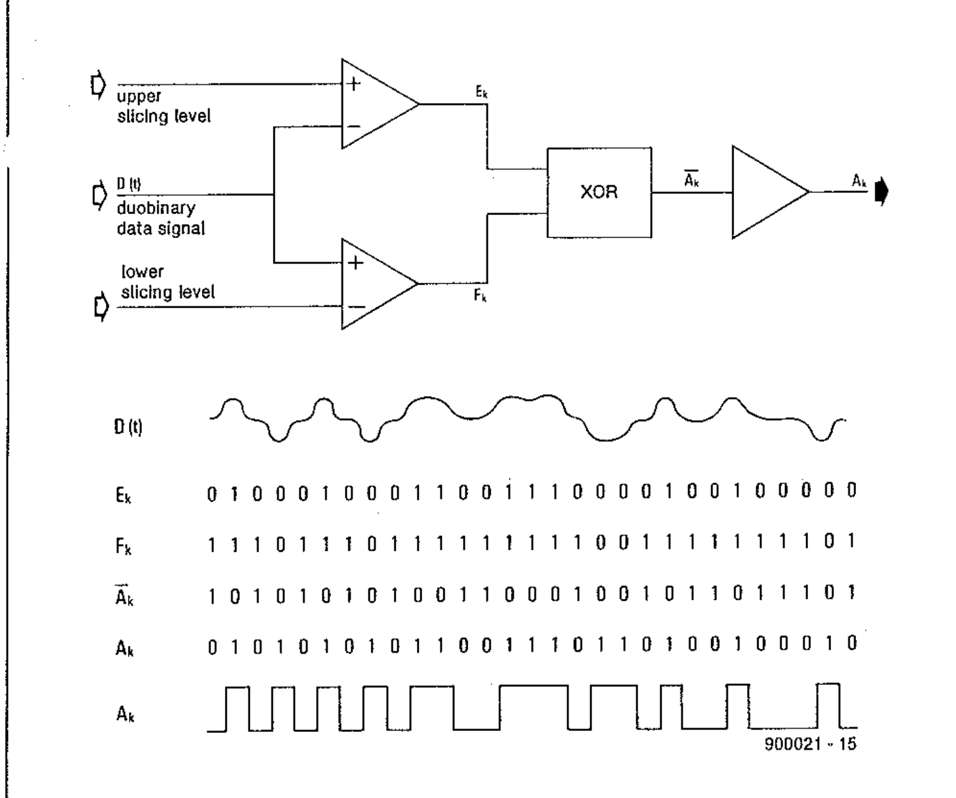

Fig. 5. Basic operation of a duobinary decoder that recovers the original sound and data bitstream marked A_k (*illustration courtesy Blaupunkt GmbH*).

Abbreviations used in this article

AM = Amplitude Modulation
BER = Bit Error Rate
BSB = British Satellite Broadcasting
CCETT = Centre Commun des Etudes Téléphonie et Télévision
EBU = European Broadcasting Union
HDTV = High Definition Television
MAC = Multiplexed Analogue Components
PAL = Phase Alternation Line
PSK = Phase Shift Keying
SECAM = Séquentiel Couleur à Memoire
VSB = Vestigial Side Band
WARC = World Administrative Radio Conference
XOR = Exclusive OR

At the receiver side: duobinary decoding

The baseband output signal of a satellite-TV receiver tuned to a MAC transmission contains the duobinary encoded signal $D_{(t)}$ received from the TV satellite. Two comparators with adjustable slicing levels may be used as shown in Fig. 4 to recover the original datastream A_k which contains the sound and data bits. In practice, the data-slicer is integrated into a MAC decoder chip such as the DMA2280 from ITT Semiconductors. To achieve a low BER, the DMA2280 allows the upper and lower slicing levels to be adjusted with the aid of an internal register.

Conclusions

Duobinary encoding and decoding are relatively simple operations that result in a significant bandwidth reduction of MAC signals transmitted by high- and medium power TV satellites. Experiments have shown that the system is highly immune to reflections and phase delays typically introduced in large cable systems.