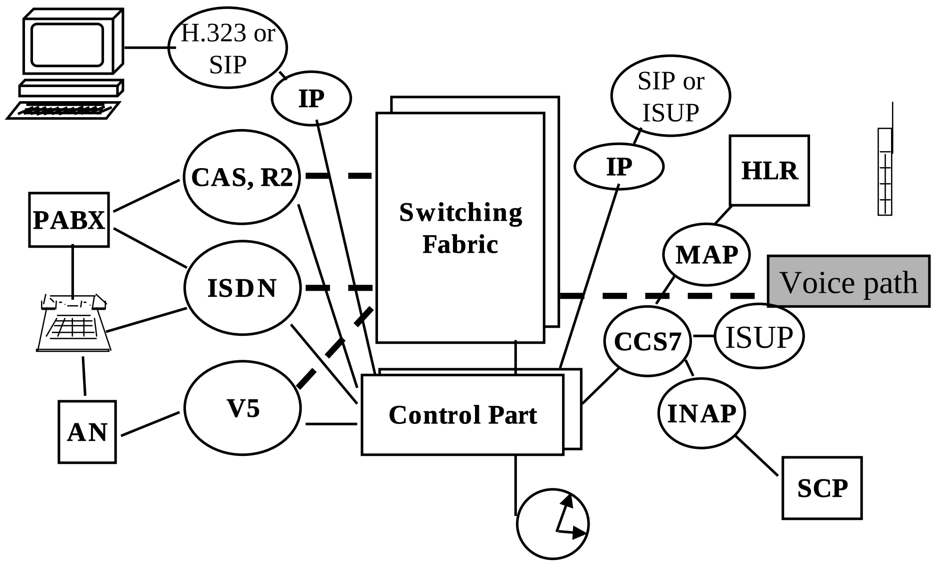
Voice path, PCM system, Line code

- ✔ **PCM ~ Pulse Code Modulation**
  - › **Sampling**
  - › **Quantizing**
    - • **Linear**
    - • **Non-linear**
  - › **Quantizing error**
- ✔ **TDM- time division multiple access**
  - › **PCM-frame structure, CRC4 -multi-frame**
  - › **PCM 30, PCM 120, PCM 480, PCM 1920 (see also course s38.118)**
  - › **PCM-line code**

Summary of course scope



Requirements for the Voice path and the Switching Fabric

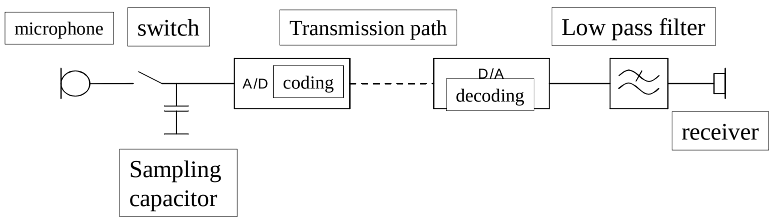
- ✔ **The Switching Fabric must understand the bits, the timeslots and the frames in the same way as the transmission systems that carry the bits**
  - › **The Fabric and the transmission systems must be synchronized**
- ✔ **Voice must be coded efficiently (what is efficient changes over time)**
- ✔ **CRC -multi-frame must enable transparent transmission (= any octet values can be sent over the network freely)**
- ✔ **An exchange must supervise voice connections:**
  - › **calls shall/should not be offered to faulty connections**
  - › **calls must sometimes be cleared from faulty connections**
  - › **detected faulty connections must be reported to far end if possible**

Sampling

- ✔ **Nyquist theorem**
  - › **If an analogue signal with limited spectrum is sampled regularly with a frequency of at least twice as high as the highest frequency component, the samples carry all the information in the original signal. The original signal can be reconstructed using a low pass filter.**
- ✔ **In voice transmission, the spectrum carried is specified to be 300 - 3400 Hz, resulting in a minimum sampling rate of 6,8 kHz.**
- ✔ **In practice, since the width of the transmission channel in an analogue system is 4kHz, in a digital system a sampling rate of 8 kHz (8000 samples/s) is used.**

Digital voice transmission

- ✔ The voice path includes a microphone, A/D-converter, D/A-converter and a loudspeaker.
- ✔ In practice, the analogue signal needs to be filtered before the conversion

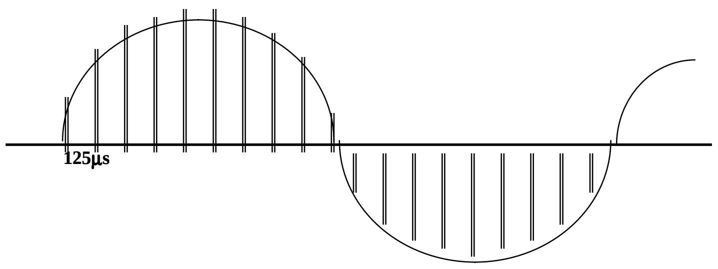


Pulse Code Modulation - PCM

- ✔ In PCM, analogue voice is digitized and thus it can be carried by digital transmission systems and switched in digital switching fabrics.
- ✔ PCM was invented in 1937 but the first real implementations became possible only using transistor technology during 1960's. This is also one of the origins of Nokia Electronics (1968) and Nokia Telecommunications.
- ✔ PCM conversion has four steps:
  - › filtering
  - › sampling
  - › quantizing
  - › coding

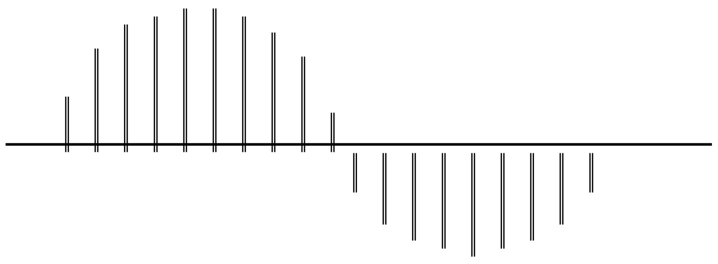
Sampling of the analogue signal

- ✔ Sampling of the analogue signal is done with a frequency of 8 kHz, I.e. with an inter-sample interval of 125 µs.
- ✔ The result is a PAM -signal



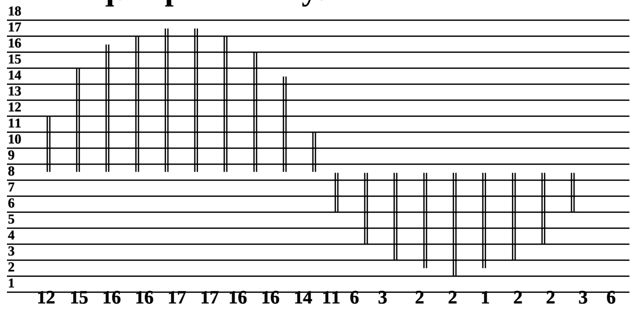
Pulse Amplitude Modulation PAM

- ✔ Sampling produces a time discrete PAM signal reflecting the amplitude of the analogue signal.
- ✔ PAM-signal is quatized producing PCM-code.



Quantizing results in approximation of the samples

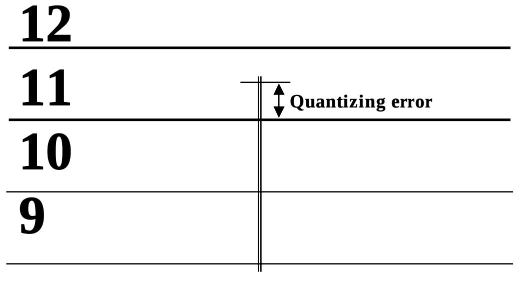
- ✔ **Real valued amplitude figures are replaced by discrete integer values.**
- ✔ **Quantizing should result in values that appear in the signal with equal probability.**



Quantizing distortion

- ✔ **Quantizing produces distortion, that is called quantizing distortion.**
- ✔ **Quantizing distortion is made by the replacement of real values by their integer approximates and at maximum can reach ½ quantizing interval.**
- ✔ **In linear quantizing the signal to distortion ratio is**

$$\boxed{S/D = 6n + 1{,}8\ \text{dB} \qquad n = \text{word length}}$$



Linear vs. non-linear

- ✔ **The result of quantizing should use signal values with equal probability.**
- ✔ **This results in minimization of distortion, because a larger number of discrete signal values falls into the most typical analogue signal value area.**
- ✔ **In a voice signal, small analogue values appear with higher probability than larger values.**

--> non-linear quantizing

Non-linearity

- ✔ **Non-linear conversion can be implemented in two ways:**
  - › **using non-linear quantizing**
  - › **using compression before linear quatizing is applied**
- ✔ **Non-linear quantizing can be implemented e.g. using a network of resistors, compression requires a non-linear amplifier.**
- ✔ **Irrespective of the method of implementation, the non-linear quantizing follows a conversion function giving the mapping of analogue signal values to integers.**
  - › **In Europe (ETSI) A-function**
  - › **In USA (ANSI) µ-function**

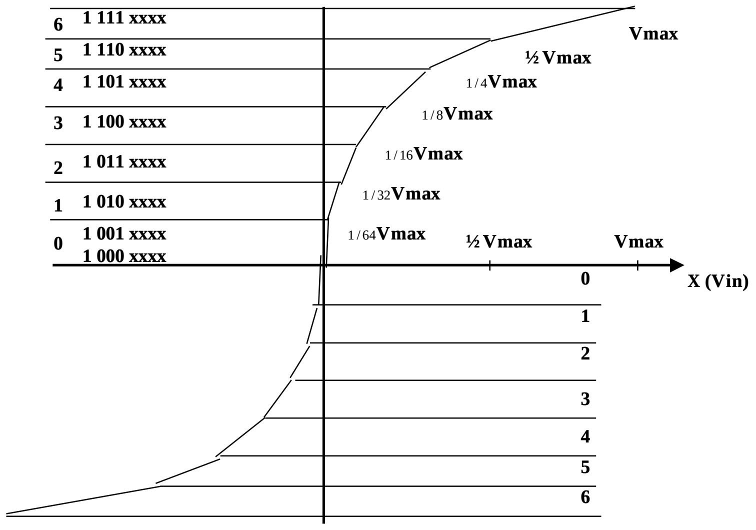
PCM-coding and quantizing

- ✔ **Accoding to ETSI specification, voice coding uses 8 bits per sample.**
  - › **bit-1 gives the polarity of the signal**
  - › **bits 2-4 give the segment of the non-linear quantizing**
  - › **bits 5-8 give the value of the discrete signal inside the segment**
- ✔ **Non-linearity follows the so called A -law**

$$\begin{cases} \dfrac{A|x|}{1+\ln(A)}, & 0 \le |x| \le \dfrac{1}{A} \\ \dfrac{1+\ln|Ax|}{1+\ln(A)}, & \dfrac{1}{A} \le |x| \le 1 \end{cases}$$

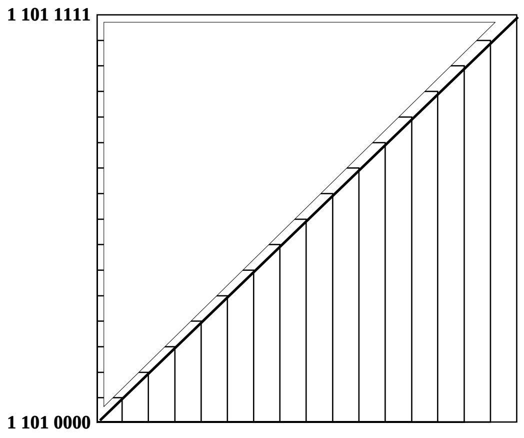
The value of A is 87,6.

Quantizing according to the A-law



Quantizing inside a Segment

✔ **In a segment quantizing is linear**



Linear vs non-linear quantizing

✔ **Linear and non-linear quantizing can be compared using the gain in signal resolution by non-linearity.**

✔ **Non-linear quantizing emphasizes small signal values, for which a gain in resolution of 24 dB is achieved.**

$$\boxed{G_{dB}=20\log V_{in}/V_{comp}}$$

PCM-hierarchy

- ✔ **PCM-hierarchy is created by overlapping time division multiplexed signal connections bit by bit. Bits become shorter.**
- ✔ **The basic speed in the hierarchy is the bitrate of a single voice channel**

S=8000Hz* 8bit = 64kbit/s

- ✔ **The following voice channel groups are difined**
  - › **30 voice channels**
  - › **120 voice channels**
  - › **480 voice channels**
  - › **1920 voice channels**

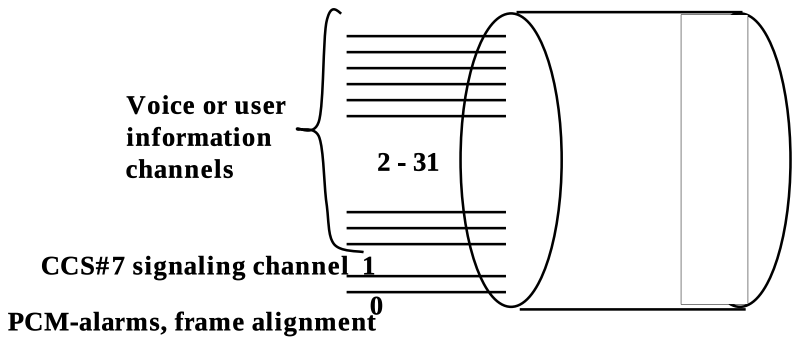
PCM 30 (E1)

- ✔ **The most common information switching and transmission format in the telecommunication network is PCM 30.**
- ✔ **PCM 30 contains:**
  - › **1 synchronization and management channel**
  - › **1 signaling channel**
  - › **30 voice channel**
- ✔ **A channel is a time slot in the PCM-frame (125µs), created by TD multiplexing.**
- ✔ **PCM 30 system carries 32 time slots, each 64kbit/s. This gives a total bit rate of 2048kbit/s.**

PCM 30 frame

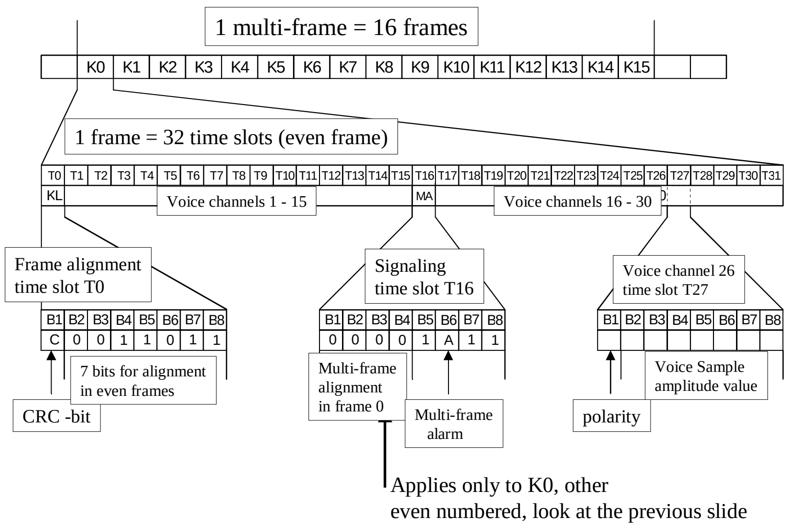
- ✔ **PCM 30 -frame contains 32 time slots**
  - › **time slot 0 is dedicated for synchronization and management information**
  - › **Time slot 16 is assigned for signaling information (CAS)**
  - › **Time slots 1-15 and 17-32 are voice or user information channels**
- ✔ **Even and odd frame structures differ**
  - › **In even numbered frames time slot 0 carries the frame alignment signal (C0 01 10 11). C is the CRC-bit (cyclic redundancy check) for ensuring the frame alignment recovery in case someone is sending X0 01 10 11 on a user information channel.**
  - › **Time slot 0 in odd frames carries alarm information. To avoid wrong frame alignment, the second bit in tsl 0 is set to the constant value of 1.**

The use of PCM time slots in the Finnish CCS#7 network

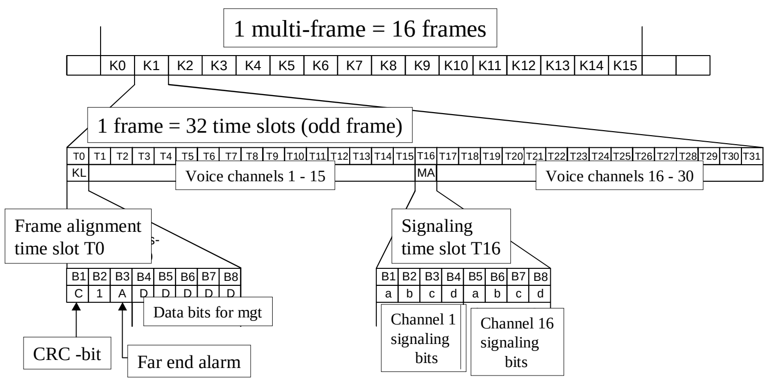

Voice or user information channels
2 - 31
CCS#7 signaling channel 1
0
PCM-alarms, frame alignment

Nowadays, tsl 16 is used for voice!

Even numbered PCM 30 -frame


1 multi-frame = 16 frames
K0 K1 K2 K3 K4 K5 K6 K7 K8 K9 K10 K11 K12 K13 K14 K15
1 frame = 32 time slots (even frame)
T0 T1 T2 T3 T4 T5 T6 T7 T8 T9 T10 T11 T12 T13 T14 T15 T16 T17 T18 T19 T20 T21 T22 T23 T24 T25 T26 T27 T28 T29 T30 T31
KL
Voice channels 1 - 15
MA
Voice channels 16 - 30
Frame alignment time slot T0
B1 B2 B3 B4 B5 B6 B7 B8
C 0 0 1 1 0 1 1
7 bits for alignment in even frames
CRC -bit
Signaling time slot T16
B1 B2 B3 B4 B5 B6 B7 B8
0 0 0 0 1 A 1 1
Multi-frame alignment in frame 0
Multi-frame alarm
Applies only to K0, other even numbered, look at the previous slide
Voice channel 26 time slot T27
B1 B2 B3 B4 B5 B6 B7 B8
Voice Sample amplitude value
polarity

PCM-frame structure (odd frame)


1 multi-frame = 16 frames
K0 K1 K2 K3 K4 K5 K6 K7 K8 K9 K10 K11 K12 K13 K14 K15
1 frame = 32 time slots (odd frame)
T0 T1 T2 T3 T4 T5 T6 T7 T8 T9 T10 T11 T12 T13 T14 T15 T16 T17 T18 T19 T20 T21 T22 T23 T24 T25 T26 T27 T28 T29 T30 T31
KL
Voice channels 1 - 15
MA
Voice channels 16 - 30
Frame alignment time slot T0
B1 B2 B3 B4 B5 B6 B7 B8
C 1 A D D D D D
Data bits for mgt
CRC -bit
Far end alarm
Signaling time slot T16
B1 B2 B3 B4 B5 B6 B7 B8
a b c d a b c d
Channel 1 signaling bits
Channel 16 signaling bits

CRC-4 calculation ensures, that the frame alignement function can not lock into a user signal of (x0011011)

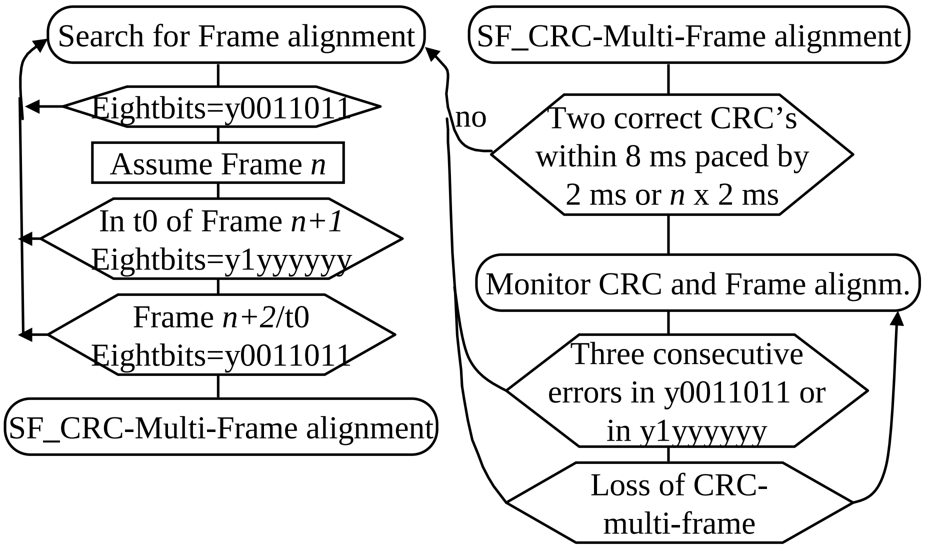
Tsl-0/bit x

		Frame nr	t0/1	t0/2	
CRC-4 multi-frame	I-half	0	C1	0	0
		1	0	1	A
		2	C2	0	0
		3	0	1	A
		4	C3	0	0
		5	1	1	A
		6	C4	0	0
		7	0	1	A
	II-half	8	C1	0	0
		9	1	1	A
		10	C2	0	0
		11	1	1	A
		12	C3	0	0
		13	E	1	A
		14	C4	0	0
		15	E	1	A

1 1 0 1 1

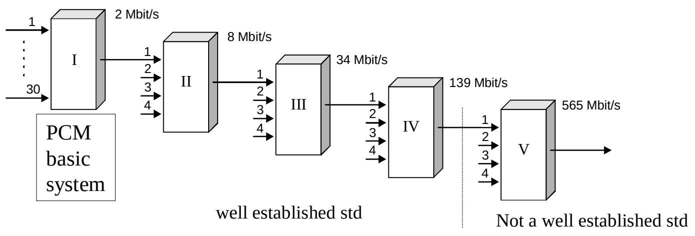
C1…C4 - CRC4 -bits
E - CRC4-error bits
001011 - CRC4 -multi-frame alignment
A - far-end alarm (t0-frame alignment lost)

E1 Frame alignement algorithm/ G.706


Search for Frame alignment
Eightbits=y0011011
Assume Frame *n*
In t0 of Frame *n+1* Eightbits=y1yyyyyy
Frame *n+2*/t0 Eightbits=y0011011
SF_CRC-Multi-Frame alignment
SF_CRC-Multi-Frame alignment
no
Two correct CRC's within 8 ms paced by 2 ms or *n* x 2 ms
Monitor CRC and Frame alignm.
Three consecutive errors in y0011011 or in y1yyyyyy
Loss of CRC-multi-frame

Higher levels in the hierarchy

- ✔ **PCM multiples**
  - › **PCM 30 (E1) 2,048Mbit/s**
  - › **PCM 120 (E2) 8,448 Mbit/s**
  - › **PCM 480 (E3) 34,368 Mbit/s**
  - › **PCM 1920 (E4) 139,264 Mbit/s**
- ✔ **Multiples are formed by multiplexing frames from four lower level connections into new higher order frames. Management overhead info is added into the higher order frame.**

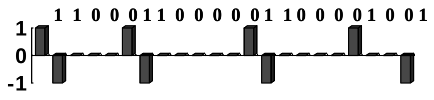


Line code in the PCM system

- ✔ **PCM-system uses bipolar transmission.**
- ✔ **Binary one is transmitted only during 50% of the cycle time.**
  - › **There is no direct current that would need to be eliminated**
  - › **Power spectrum is concentrated around ½ bit rate**
- ✔ **Two alternative line codes are used in the PCM system:**
  - › **AMI - Alternate Mark Inversion**
  - › **HDB3 - High Density Binary 3**

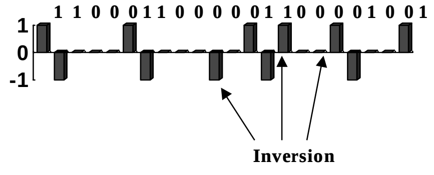
Alternate Mark Inversion - AMI

- ✔ **In the AMI-code**
  - › **Binary one changes polarity at each occurrence**
  - › **Binary zero is no signal on the line**
- ✔ **Weakness is in loosing bit sync in case of long series of zeroes.**



High Density Binary 3 - HDB3

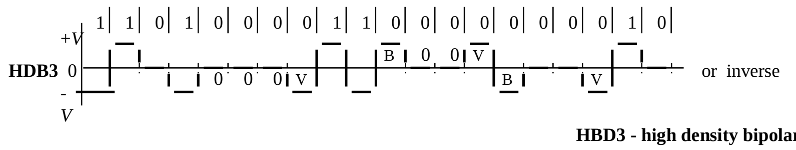
- ✔ **In HDB3 code**
  - › **Binary one changes polarity as in the AMI-code**
  - › **Binary zero**
    - **First zero is replaced by "one" if in the previous group, an inversion was used**
    - **Second and third zeroes are no signal**
    - **The fourth consecutive zero raises an inversion - I.e. violation of the code, I.e. a pulse with the same polarity as the previous one or "one" is transmitted**



HDB3 mapping rules

Nrof B pulses after previous violation	Polarity of last B pulse	Line code	Representation of Line code
Odd	Negative (-)	000-	000V
Odd	Positive (+)	000+	000V
Even	Negative (-)	+00+	B00V
Even	Positive (+)	-00-	B00V

B - bipolar pulse



HBD3 - high density bipolar 3
V - violation
B - balance

HDBN -receiver initializes by detecting a violation and starts decoding groups of N+1 -bits

A tentative algorithm:
Reconstructed from the text of the spec.

HDB3 -receiver:

Initiali-zation

- **Interpret 0=0, +/- = 1, count pulses (odd, even)**
- **Until an AMI code violation is found in the signal = two consecutive + or - pulses,**
- **=>interpret odd& previous 000V = 0000, even& prev. B00V =0000 =>next bit starts a groups of 4 bits**

- **count the nrof pulses**
  - **0 --> interpret as 0**
  - **+ --> interpret as 1 (increment nrof-pulses)**
  - **- --> interpret as 1 (increment nrof-pulses)**
- **nrof pulses = odd & 000V**
  interpret = 0000
- **nrof-pulses=even & B00V**
  interpret = 0000