L04 - TDM Techniques

TDM Techniques

Time Division Multiplexing (synchronous, statistical)
Digital Voice Transmission, PDH, SDH

Agenda

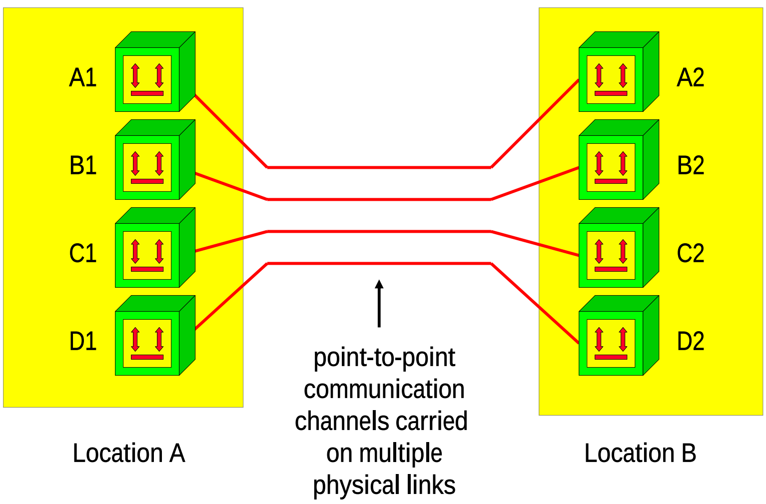
- **Introduction**
- **Synchronous (Deterministic) TDM**
- **Asynchronous (Statistical) TDM**
- **Digital Voice Transmission**
- **E1 Framing**
- **T1 Framing**

 TDM Techniques, v4.7 2

Introduction

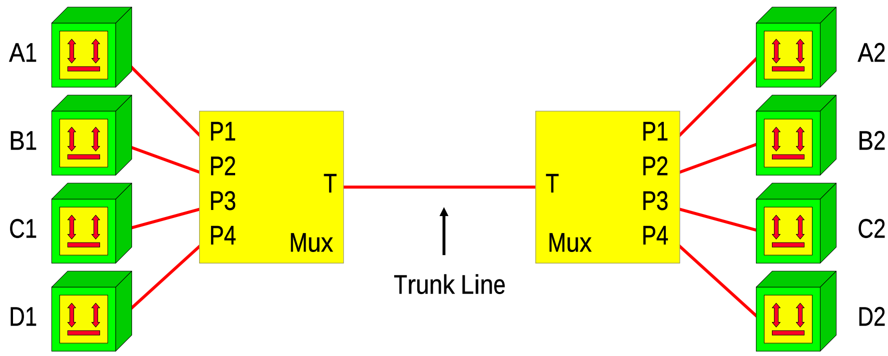
- **line protocol techniques (data link procedures)**
  - were developed for communication between two devices on one physical point-to-point link
  - bandwidth of physical link is used exclusively by the two stations
- **in case multiple communication channels are necessary between two locations**
  - multiple physical point-to-point are needed
  - expensive solution
- **in order to use one physical link for multiple channels**
  - multiplexing techniques were developed

Point-To-Point Channels



Multiplexing / Demultiplexing

- **multiplexer is a device**
  - which can take a number of input channels and, by interleaving them, output them as one data stream on one physical trunk line



 TDM Techniques, v4.7 5

Time Division Multiplexing (TDM)

- **time division multiplexer**
  - allocates each input channel a period of time or timeslot
  - controls bandwidth of trunk line among input channels
- **individual time slots**
  - are assembled into frames to form a single high-speed digital data stream
- **available transmission capacity of the trunk**
  - is time shared between various channels
- **at the destination demultiplexer reconstructs**
  - individual channel data streams

 TDM Techniques, v4.7 6

Types of TDM

- **depending on timing behavior two methods**
  - synchronous TDM
    - timeslots have constant length (capacity) and can be used in a synchronous, periodical manner
  - asynchronous (statistical) TDM
    - timeslots have variable length and are used on demand (depending on the statistics of channel communication)

Synchronous TDM Standards

- **TDM framing on the trunk line**
  - can be vendor dependent
    - proprietary TDM products
  - can be standard based
- **two main architectures for standardizing synchronous TDM for trunk lines**
  - **PDH - Plesiochronous Digital Hierarchy**
    - e.g. E1 (2Mbit/s), E3 (34Mbit/s), E4, T1 (1,544Mbit/s), T3
  - **SDH - Synchronous Digital Hierarchy**
    - e.g. STM-1 (155Mbit/s), STM-4 (622Mbit/s), STM-16

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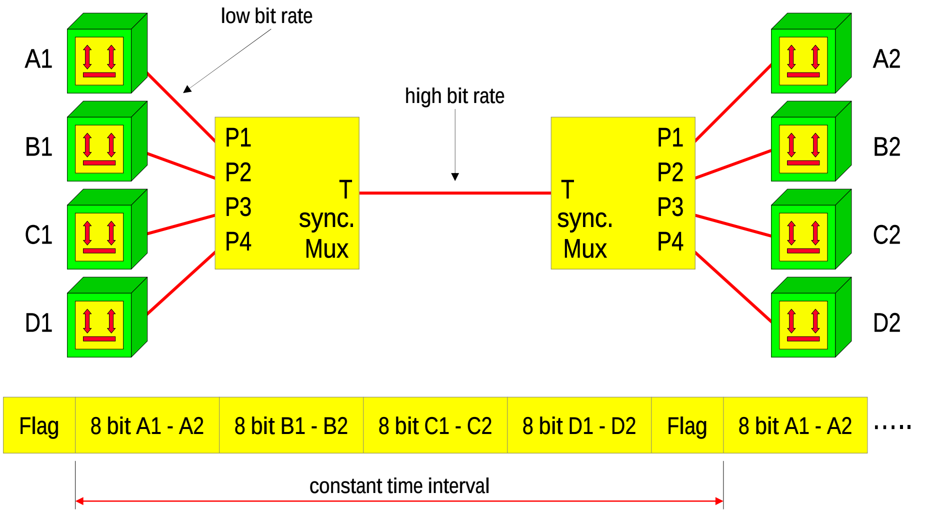
Synchronous Time Division Multiplexing

- **synchronous TDM**
  - periodically generates a frame consisting of a constant number of timeslots each timeslot of constant length
  - timeslots can be identified by position in the frame
    - timeslot 0, timeslot 1, ...
  - frame synchronization achieved by extra flag field
- **every input channel is assigned**
  - a reserved timeslot
  - e.g. timeslot numbers refer to port numbers of a multiplexer
    - traffic of port P1 in timeslot 1 for A1- A2 channel
    - traffic of port P2 in timeslot 2 for B1- B2 channel
    - ......

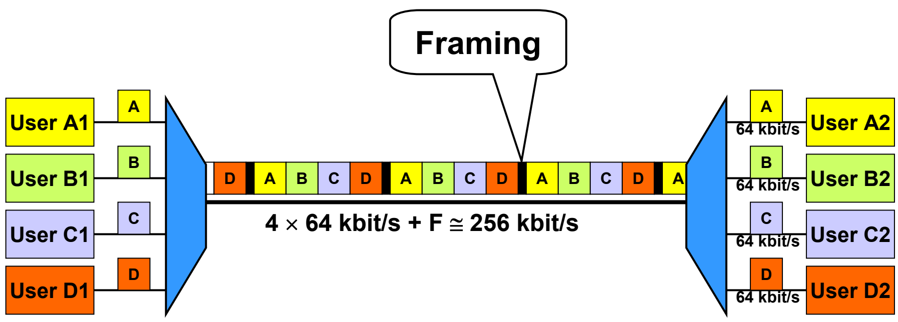
Synchronous Time Division Multiplexing

Implicit addressing given by the position of a timeslot in the frame



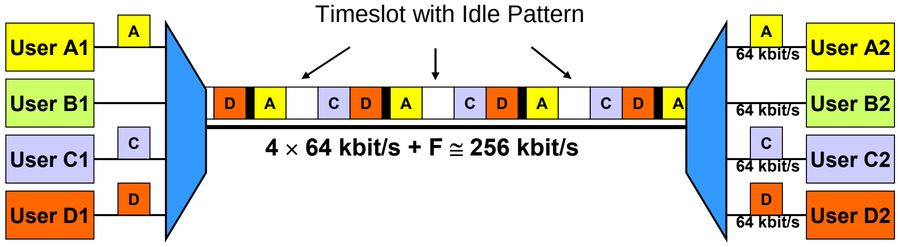
Trunk Speed with Synchronous TDM

Implicit addressing given by the position of a timeslot in the frame



- **Trunk speed = Number of slots × User access rate**
- **Each user gets a constant timeslot of the trunk**

Idle Timeslots with Synchronous TDM



- If a communication channel has nothing to transmit
- -> Idle timeslots -> Waste of bandwidth

Advantages

- **compared to pure point-to-point physical links**
  - synchronous multiplexing adds only minimal delays
    - time necessary to packetize and depacketize a byte
    - transmission/propagation delay on trunk
- **the delay for transporting a byte is <u>constant</u>**
- **the time between two bytes to be transported is constant**
  - hence optimal for synchronous transmission requirements like traditional digital voice
- **any line protocol could be used between devices**
  - method is <u>protocol-transparent</u>
- **to endsystems**
  - channel looks like a single physical point-to-point line

Disadvantages

- **bitrate on trunk line T**
  - sum of all port bitrates (P1-P4) plus frame synchronization (flag)
  - high bitrate is required
  - hence expensive
- **if no data is to be sent on a channel**
  - special idle pattern will be inserted by the multiplexer in that particular timeslot
  - waste of bandwidth of trunk line
- **asynchronous (statistic) time division multiplex avoids both disadvantages**
  - making use of communication statistics between devices

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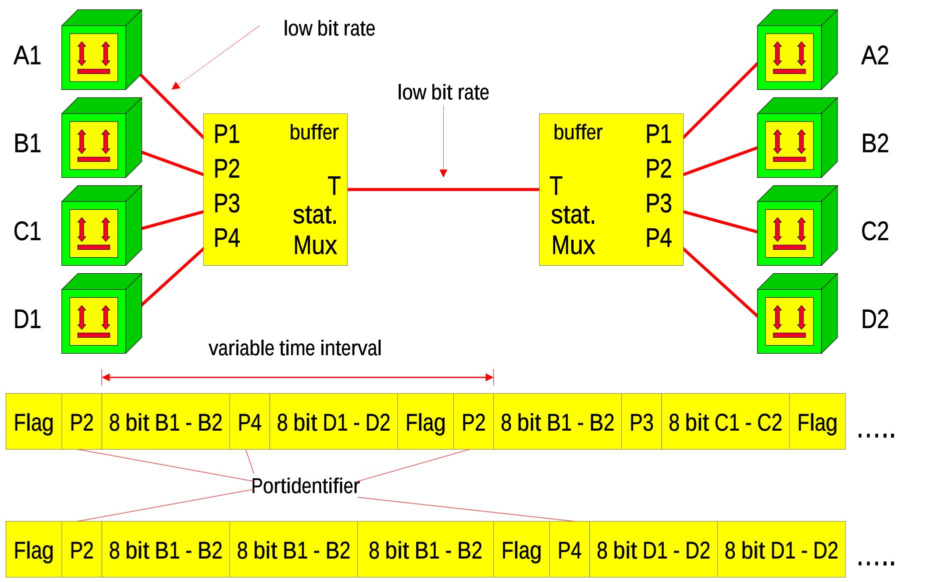
- **<u>Asynchronous (Statistical) TDM</u>**
- **Voice Transmission**
- **E1 Framing**
- **T1 Framing**

Asynchronous Time Division Multiplexing

- **usually devices communicate in a statistical manner**
  - not all devices have data to transmit at the same time
- **therefore it is sufficient**
  - to calculate necessary bitrate of the multiplexer trunk line according to the average bitrates caused by device communication
- **if devices transmit simultaneously**
  - only one channel can occupy trunk line
  - data must be buffered inside multiplexer until trunk is available again (store and forward principle)
  - statistics must guarantee that trunk will not be monopolized by a single channel

Asynchronous Time Division Multiplexing



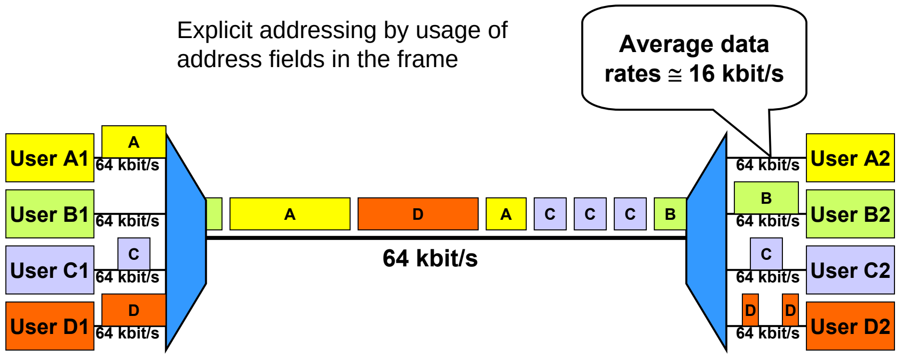
ATDM Operation

- **multiplexer only generates a transmission frame**
  - if data octets are present at input ports
- **source of data**
  - must be explicitly identified in transmission frames
  - addressing
- **reason for addressing**
  - there exists no constant relationship between timeslot and portnumber as with synchronous TDM
    - Note: addressing in synchronous TDM is implicit by recognizing the flag of the frame and hence the position of a certain timeslot
- **port identifier**
  - is used as address of source and sent across the trunk

ATDM Operation / Facts

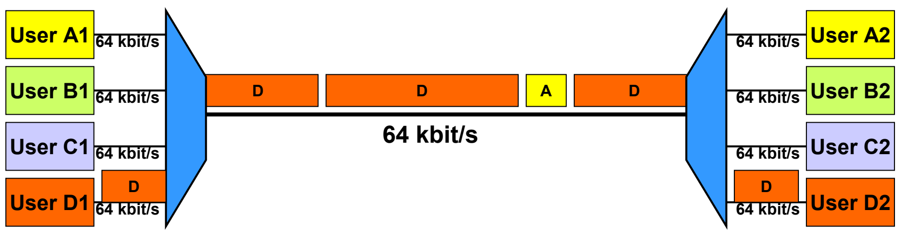
- **transmission frame can be assembled using**
  - either a single channel octet by frame
    - suitable for character oriented terminal sessions
  - or multiple channel octets per frame
    - suitable for block oriented computer sessions
- **in case of congestion**
  - buffering causes additional delays compared to synchronous TDM
- **delays are variable because of statistical behavior**
  - hence not optimal for synchronous transmission requirements like traditional digital voice
  - sufficient for transmission requirements of bursty data transfers

Asynchronous / Statistical TDM



- Trunk speed dimensioned for average usage
- Each user can send packets whenever he wants
- Buffering necessary if trunk already occupied

Asynchronous / Statistical TDM



- If other users are silent, one user can fully utilize his access rate

ATDM Facts

- **ATDM can be used protocol transparent**
  - however in case of buffer overflow transmission errors will be seen by devices
    - FCS errors
- **to avoid FCS errors a kind of <u>flow control between multiplexer and device (end system)</u> should be used**
  - which is a new element in data communication methods
  - this is different from flow control between end systems learned so far in module about line protocols
  - examples for flow control
    - HW flow control based on handshake signals (e.g. RTS, CTS)
    - SW flow control (e.g. XON/XOFF)
    - Protocol based flow control such as known in connection oriented line protocols like HDLC (e.g. RR and RNR)
  - end system and ADTM have to speak the same protocol language

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- **<u>Voice Transmission</u>**
- **E1 Framing**
- **T1 Framing**

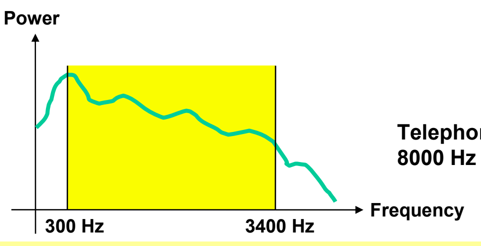
Voice Transmission

- **digital voice transmission**
  - based on Nyquist´s Theorem
  - analogous voice can be digitized using pulse-code-modulation (PCM) technique requiring a 64kbit/s digital channel
    - voice is sampled every 125usec (8000 times per second)
    - every sample is encoded in 8 bits
  - used nowadays in the backbone of our telephone network
  - today analogous transmission only between home and local office -> so called local loop
- **synchronous TDM**
  - originated from digital voice transmission

 TDM Techniques, v4.7 25

Sampling of Voice

- **Nyquist´s Theorem**
  - any analogue signal with limited bandwidth f_B can be sampled and reconstructed properly when the sampling frequency is $2 \cdot f_B$
  - transmission of sampling pulses allows reconstruction of original analogous signal
  - sampling pulses are quantized resulting in binary code word which is actually transmitted

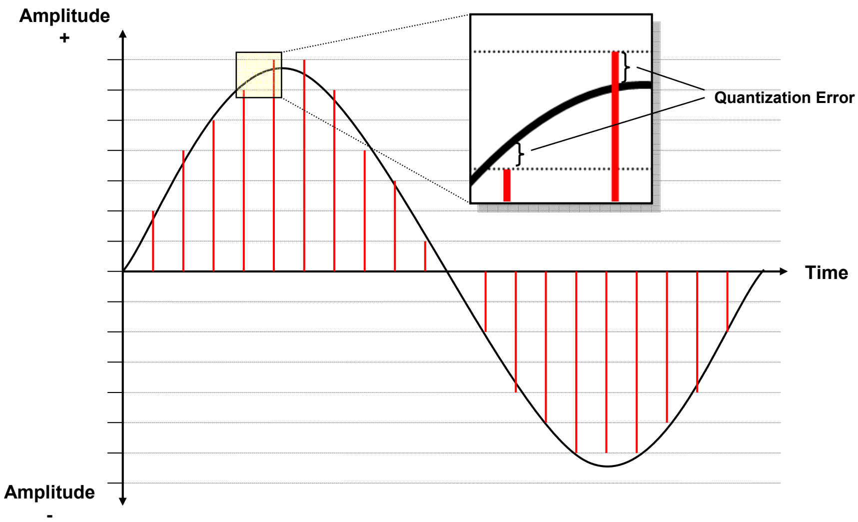


$$R = 2 * B * \log_2 V$$

Telephone channel: 300-3400 Hz
8000 Hz x 8 bit resolution = 64 kbit/s

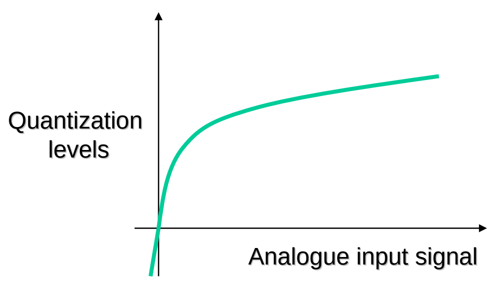
 TDM Techniques, v4.7 26

Linear Quantization

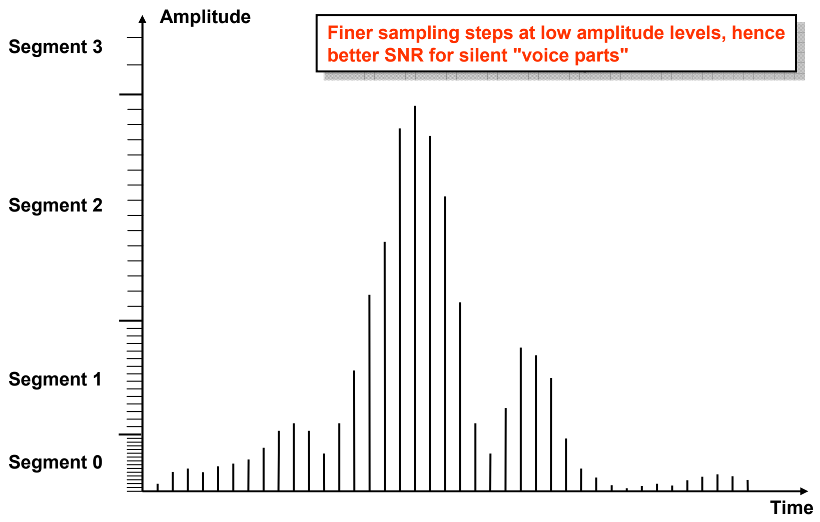


Improving SNR (Signal Noise Ratio)

- **to improve the SNR of speech signals**
  - lower amplitudes receive a finer resolution than greater amplitudes
- **a nonlinear function (logarithmic) is used for quantization**
  - USA: μ-law (Bell)
  - Europe: A-law (ITU)

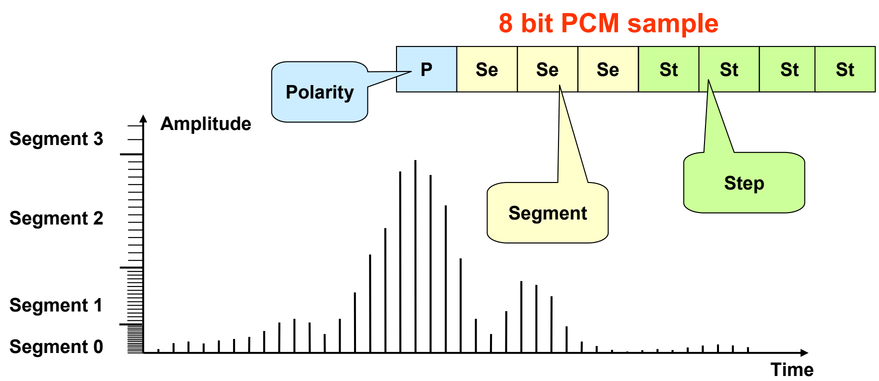


Log. Quantization



Encoding (PCM)

- Putting digital values in a defined form for transmission



Voice Compression

- **Waveform Coders**
  - Non-linear approximation of analog waveform
  - PCM (no compression), ADPCM
- **Vocoders**
  - speech is analyzed and compared to a codebook
  - only codebook values are transmitted and speed synthesizer at the receiver
- **Hybrid coders**
  - Combination of waveform coders and vocoders
  - 4.8 kbps to 16 kbps
  - Used for mobile phones
  - CELP, GSM

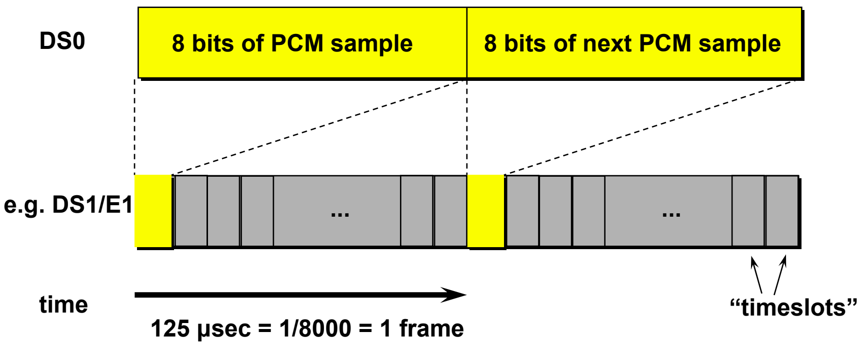
Standardized Codec's

- PCM
  - G.711 (64 kbps)
- Adaptive Differential Pulse Code Modulation (ADPCM)
  - only the difference from one sample pulse to the next will be transmitted
  - fewer bits used for encoding the difference value
  - G.726 (16, 24, 32, 40 kbps)
- Low Delay Code Excited Linear Predictor (LD-CELP)
  - G.728 (16 kbps)
- Conjugate Structure Algebraic Code Excited Linear Predictor (CS-ACELP)
  - G.729 (8 kbps)
- Dual Rate Speech Coding Standard G.723
  - is the basic standard for voice transmission in IP networks
  - basis is the CELP-Technique of GSM
  - uses minimal data rate of 5,3K at fair quality or 6,3K with good quality

Digital voice channel

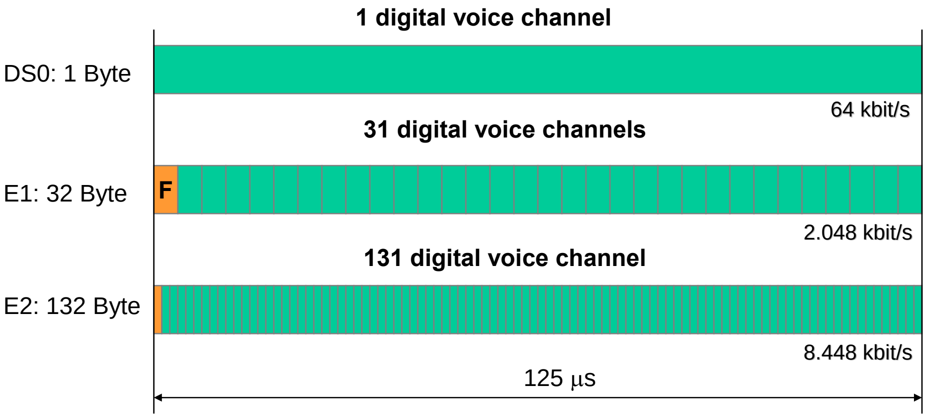
- **DS0 = Digital Signal, Level 0**
  - 1 timeslot in multiplexing frames
- **Base for hierarchical digital communication systems**
- **Equals one PCM coded voice channel**
  - 64 kbit/s
- **Each samples (byte) must arrive within 125 μs**
  - To receive 8000 samples (bytes) per second
  - Higher order frames must ensure the same byte-rate per user(!)

Multiplexing Basics



- **frame rate is always 8000 frame per second at all levels of the hierarchy**
- **byte interleaved multiplexing**

Multiplexing Basics



– note: DS0 and higher rates can be used for any transport digital information -> data transmission

Multiplexing Hierarchies

- **why hierarchy and standardization?**
  - only a hierarchical digital multiplexing infrastructure which is standardized
    - can connect millions of (low speed) customers across the city/country/world
- **two main architectures**
  - PDH - plesiochronous digital hierarchy
    - plesio means nearly synchronous, clock differences are compensated by bit stuffing techniques / overhead bits
    - PDH is still used for low-speed lines
  - SDH - synchronous digital hierarchy
    - overcomes deficits of PDH
    - in North America SONET is used
    - telecommunication backbones move very quickly to SONET/SDH

PDH Hierarchy

North America / ANSI

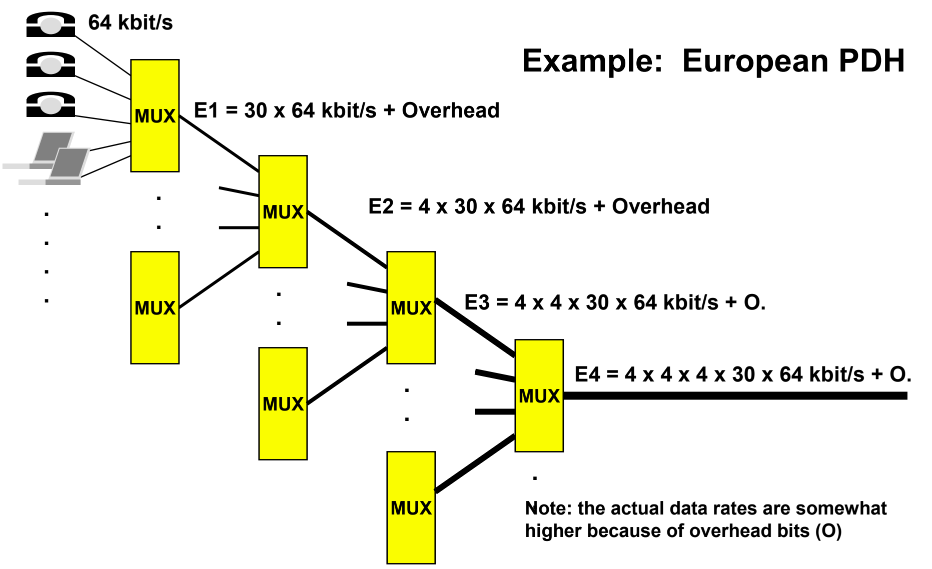
Signal	Carrier	Channels	Mbit/s
DS0		1	0.064
DS1	T1	24	1.544
DS1C	T1C	48	3.152
DS2	T2	96	6.312
DS3	T3	672	44.736
DS4	T4	4032	274.176

Europe / ITU

Signal	Carrier	Channels	Mbit/s
DS0	"E0"	1	0.064
CEPT-1	E1	32	2.048
CEPT-2	E2	128	8.448
CEPT-3	E3	512	34.368
CEPT-4	E4	2048	139.264
CEPT-5	E5	8192	565.148

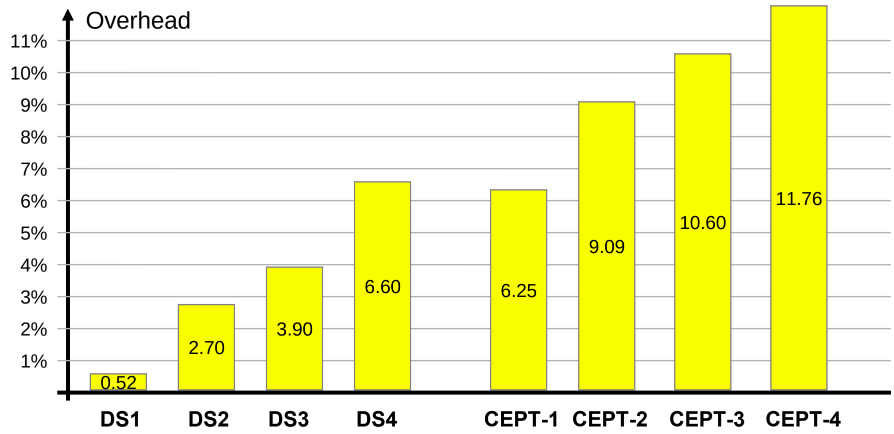
- Incompatible MUX rates
- Different signalling schemes
- Different overhead
- μ-law versus A-law

Digital Hierarchy of Multiplexers



PDH Limitations

- PDH overhead increases dramatically with high bit rates



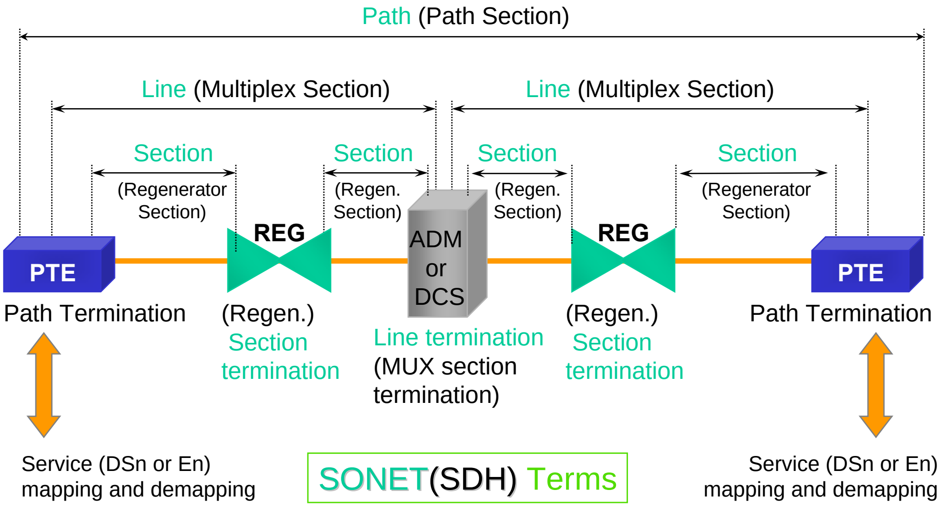
Reasons for SONET/SDH Development

- **Incompatible PDH standards !!!**
- **PDH does not scale to very high bit rates**
  - Increasing overhead
  - Various multiplexing procedures
  - Switching of channels requires demultiplexing first
- **Demand for a true synchronous network**
  - No pulse stuffing between higher MUX levels
  - Phase shifts are compensated by floating payload and pointer technique
- **Demand for add-drop MUXes and ring topologies**

SDH History

- After divestiture of AT&T
  - Many companies -> many proprietary solutions for PDH successor technology
- In 1984 ECSA (Exchange Carriers Standards Association) started on SONET
  - Goal: one common standard
  - Tuned to carry US PDH payloads
- In 1986 CCITT became interested in SONET
  - Created SDH as a superset
  - Tuned to carry European PDH payloads including E4 (140 Mbit/s)
- SDH is a world standard
  - SONET is subset of SDH
- Originally designed for fiber optics

Network Structure



SONET/SDH Line Rates

SONET Optical Levels	SONET Electrical Level	Line Rates Mbit/s	SDH Levels
OC-1	STS-1	51.84	STM-0
OC-3	STS-3	155.52	STM-1
OC-9	STS-9	466.56	STM-3
OC-12	STS-12	622.08	STM-4
OC-18	STS-18	933.12	STM-6
OC-24	STS-24	1244.16	STM-8
OC-36	STS-36	1866.24	STM-12
OC-48	STS-48	2488.32	STM-16
OC-96	STS-96	4976.64	STM-32
OC-192	STS-192	9953.28	STM-64
OC-768	STS-768	39813.12	STM-256

Defined but later removed, and only the multiples by four were left!

(Coming soon)

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- Asynchronous (Statistical) TDM
- Voice Transmission
- E1 Framing
- T1 Framing

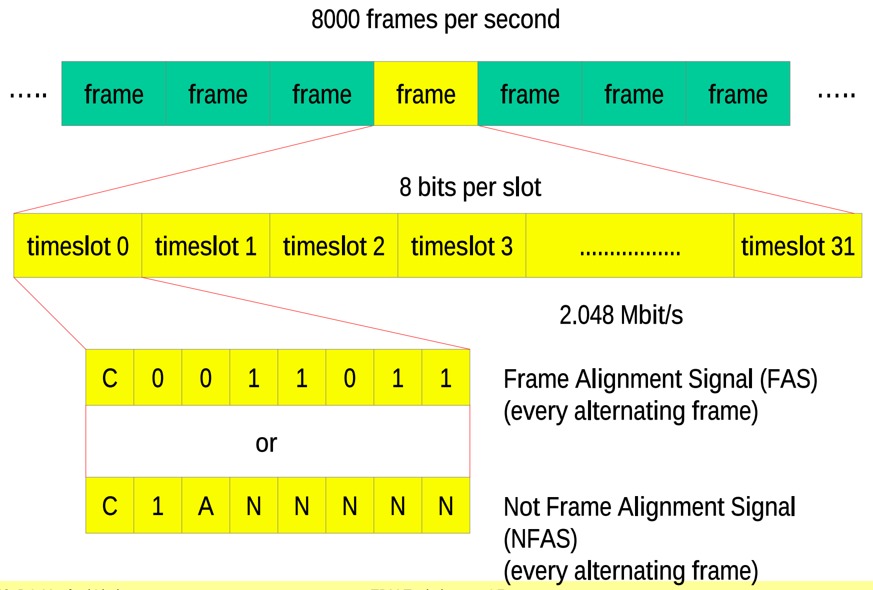
E1 Basics

- **synchronous TDM**
  - originated from digital voice transmission
  - Nyquist´s Theorem
  - analogous voice can be digitized using pulse-code-modulation (PCM) technique requiring a 64kbit/s digital channel
    - voice is sampled every 125usec (8000 times per second)
    - every sample is encoded in 8 bits
  - CEPT standardizes E1
    - as part of European channelized framing structure for PCM transmission (PDH)
    - E1 (2 Mbit/s), E2 (8 Mbit/s), E3 (34Mbit/s), E4 (139Mbit/s)
    - relevant standards G.703, G.704, G.732

E1 Framing

- **G.704 specifies framing structures for different interface rates**
  - E1 is specified at interface rate of 2.048Mbit/s
  - 32 timeslots per frame numbered 0-31
  - timeslot 0 for frame synchronization
    - allows distinction of frames and timeslots within frames
  - one timeslot can carry 8 bits
  - frame length 256 bits
  - frame repetition rate is 8000 Hz
  - 32 x 8 x 8000 = 2.048 Mbit/s
  - timeslot 16 can be used for signaling

E1 Frame Structure



E1 Frame Structure

- **every second frame**
  - timeslot 0 contains FAS used for frame synchronization
- **C (CRC) bit**
  - is part of an optional 4-bit CRC sequence
  - provides frame checking and multiframe synchronization
- **A (Alarm Indication) bit**
  - so called Yellow (remote) alarm
  - used to signal loss of signal (LOS) or out of frame (OOF) condition to the far end
- **N (National) bits**
  - reserved for future use

CRC Multiframe Structure Timeslot 0

	timeslot 0		timeslot 1 .... timeslot 31
frame 0	C1	FAS	semimultiframe 1
frame 1	0	NFAS	
frame 2	C2	FAS	
frame 3	0	NFAS	
frame 4	C3	FAS	
frame 5	1	NFAS	
frame 6	C4	FAS	
frame 7	0	NFAS	
frame 8	C1	FAS	semimultiframe 2
frame 9	1	NFAS	
frame 10	C2	FAS	
frame 11	1	NFAS	
frame 12	C3	FAS	
frame 13	Si	NFAS	
frame 14	C4	FAS	
frame 15	Si	NFAS	

CRC Multiframe Sync - bits: 0 0 1 0 1 1

CRC Multiframe Structure

- **CRC check is optional feature**
  - 16 frames are combined to a multiframe
  - start of multiframe can be detected by CRC Multiframe Sync bits
  - semimultiframe 2 contains four CRC bits, which were calculated over semimultiframe 1
  - Si bits
    - are used to report CRC errors to the far end

E1 Signaling Timeslot 16

- **E1 framing is often used to connect PBX (Private Branch Exchanges) via leased line**
- **timeslot 16 can carry out-band signaling information between PBX´s**
- **two types**
  - **Common Channel Signaling (CCS)**
    - transparent channel (capacity 64kbit/s) for signaling protocols like DPNSS, CorNet, QSIG
  - **Channel Associated Signaling (CAS)**
    - additional CAS multiframe structure
    - provides 4 bit signaling information per timeslot every 16th frame
    - 30 independent signaling channels (capacity 2kbit/s per channel)

CAS Multiframe Structure Timeslot 16

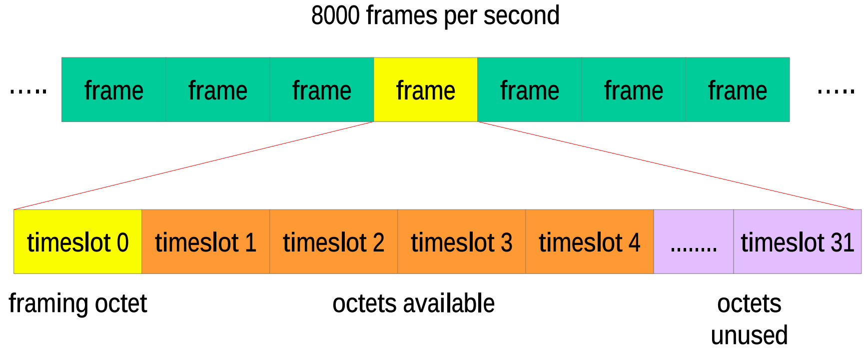
timeslots 0-15	timeslot 16		timeslots17-31
frame 0	0 0 0 0	X Y X X	
frame 1	A B (01) C D	A B (17) C D	
frame 2	A B (02) C D	A B (18) C D	
frame 3	A B (03) C D	A B (19) C D	
frame 4	A B (04) C D	A B (20) C D	
frame 5	A B (05) C D	A B (21) C D	
frame 6	A B (06) C D	A B (22) C D	
frame 7	A B (07) C D	A B (23) C D	
frame 8	A B (08) C D	A B (24) C D	
frame 9	A B (09) C D	A B (25) C D	
frame 10	A B (10) C D	A B (26) C D	
frame 11	A B (11) C D	A B (27) C D	
frame 12	A B (12) C D	A B (28) C D	
frame 13	A B (13) C D	A B (29) C D	
frame 14	A B (14) C D	A B (30) C D	
frame 15	A B (15) C D	A B (31) C D	

0000
CAS Multiframe Alignment signal

A B C D are signaling bits for the timeslot indicated in ( )

Y is Multiframe Yellow alarm bit to signal a Loss of Multiframe (LOM)
X bits not used (set to 1)

Fractional E1



Only a portion or a fraction in every frame is available for data octets (example above is called E1/4). This is typically done for financial reasons since service provide charges a lower price than for the full capacity. However the fractional E1 line still has 8000 frames per second, the total bit rate of the line (2.048 Mbit/s) is absolutely unchanged.

E1 Operational and Physical Aspects

- **G.732 specifies**
  - characteristics of PCM multiplex equipment operating at 2.048Mbit/s
    - based on frame structure G.704
    - encoding law when converting analogue to digital to be A-law
    - procedures for loss and recovery of frame alignment, for fault conditions and consequent actions, for acceptable jitter levels
- **G.703 specifies**
  - electrical and physical characteristics
    - 75 ohm coax, unbalanced
    - 120 ohm twisted pair, balanced
  - encoding
    - HDB3

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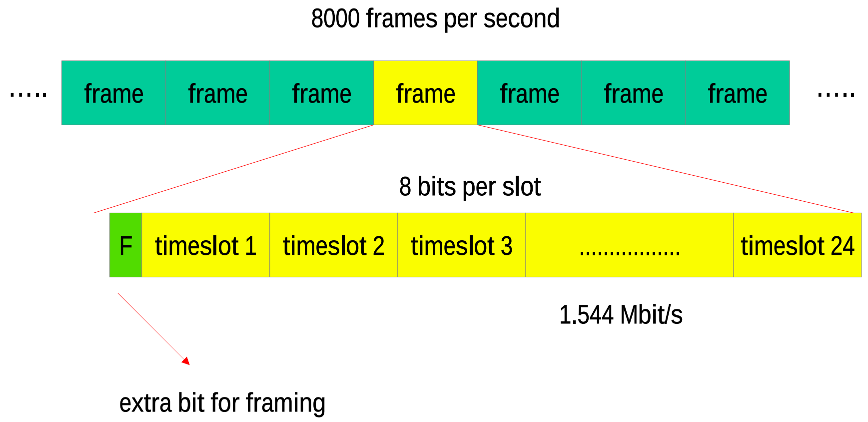
T1 Basics

- **T1 is North American channelized framing structure for PCM transmission**
  - synchronous TDM originated from digital voice transmission
  - a 64kbit/s digital channel used for carrying PCM encoded voice is called DS0
  - DS0 is basic element with lowest bitrate of North American PDH (plesiosynchronous digital hierarchy)
    - DS0, DS1 (T1), DS2, DS3 (45MBit/s), DS4 (274Mbit/s)
  - encoding and physics:
    - AMI or B8ZS (Bipolar 8 Zero bit Suppression)
    - 100 ohm, twisted pair

T1 Framing

- **T1 frame**
  - 24 timeslots per frame numbered 1-24
  - one extra bit for framing
  - one timeslot can carry 8 bits
  - frame length 193 bits
  - frame repetition rate is 8000 Hz
  - 24 x 8 x 8000 = 1.544 Mbit/s

T1 Basic Frame Structure

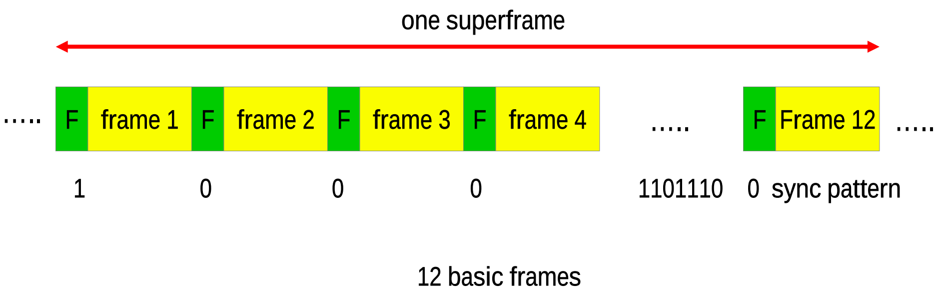


Superframe

- **one framing bit is not sufficient for frame synchronization**
  - framing bits of consecutive frames are combined to form a multiframe synchronization pattern
  - multiframe structure is called superframe

- **D4 format**
  - 12 frames are combined to one superframe (SF)
  - 12 consecutive framing bits are 100011011100 (1200 bits/s used for synchronization)

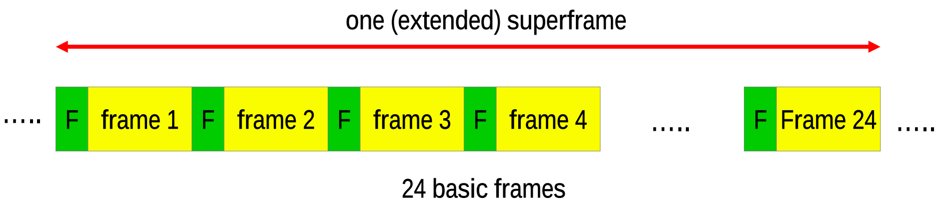
D4 Format



Extended Superframe

- **ESF format**
  - 24 frames are combined to one extended superframe (ESF)
  - 6 framing bits (2000bit/s) are used for synchronization in frames 4, 8, 12, 16, 20, 24 (pattern 001011)
  - 6 framing bits (2000 bit/s) may be used for CRC error checking in frames 2, 6, 10, 14, 18, 22
  - 12 framing bits (4000 bit/s) may be used for a diagnostic channel in all odd numbered frames

ESF Format



sync pattern 001011 in frames 4, 8, 12, 16, 20, 24
six CRC bits in frames 2, 6, 10, 14, 18, 22
diagnostic bits in frame 1, 3, 5, 7, 9, 11, 13, 15, 17, 19, 21, 23

T1 Signaling

- **T1 framing is often used to connect PBX (Private Branch Exchanges) via leased line and hence signaling information between PBX´s must be exchanged**
- **T1 defines no reserved timeslot for signaling**
- **for Channel Associated Signaling (CAS)**
  - robbed bit signaling is used
    - signaling information is transmitted by robbing certain bits, which are normally used for data
    - signaling is placed in the LSB of every time slot in the 6th and 12th frame of every D4 superframe (A, B)
    - signaling is placed in the LSB of every time slot in the 6th, 12th 18th and 24th frame of every ESF superframe (A, B, C, D)

Robbed Bit Signaling D4

	timeslot 1		....	timeslot 24	
frame 1	Data			Data	
frame 2	Data			Data	
frame 3	Data			Data	
frame 4	Data			Data	
frame 5	Data			Data	
frame 6	Data	A		Data	A
frame 7	Data			Data	
frame 8	Data			Data	
frame 9	Data			Data	
frame 10	Data			Data	
frame 11	Data			Data	
frame 12	Data	B		Data	B

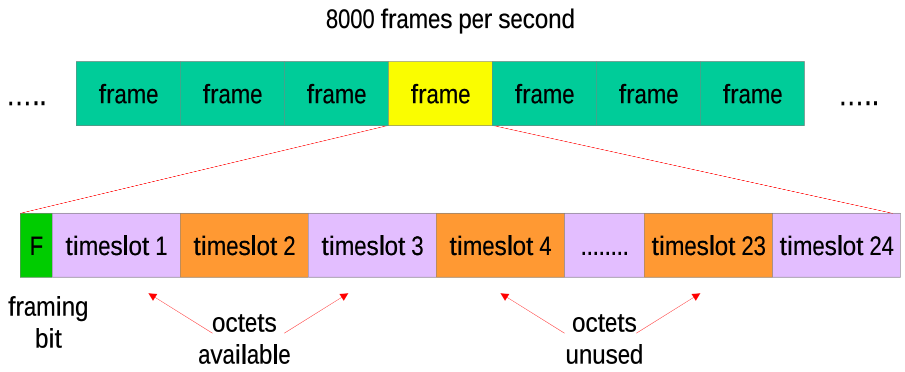
Robbed Bit Signaling ESF

	timeslot 1		....	timeslot 24	
frame 1	Data			Data	
frame 2	Data			Data	
....					
frame 6	Data	A		Data	A
....					
frame 12	Data	B		Data	B
....					
frame 18	Data	C		Data	C
....					
frame 22	Data			Data	
frame 23	Data			Data	
frame 24	Data	D		Data	D

Robbed Bit Signaling

- **Robbed Bit Signaling**
  - in case of transmitting PCM samples stealing one least significant bit every 6th frame has no severe influence on speech reconstruction
- **T1 system which uses this technique**
  - cannot carry 24 transparent data channels of 64kbit/s each
  - only n x 56 kbit/s data channels are possible
- **Common Channel Signaling (CCS)**
  - can be used in the same way like E1
  - e.g. timeslot 24 is used as transparent signaling channel

Fractional T1



Only a portion or a fraction in every frame is available for data octets (example above is called T1/12).
However the fractional T1 line still has 8000 frames per second, the total bit rate of the line (1.544 Mbit/s) is absolutely unchanged.