Data Communications and Network Technologies

“E1 and Switching”

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Data Sources

- Symbols: decimal and binary number systems, digits, alphabets, documents, images, graphics, tables, video, keyboard keys,…
  - Decimal Numbers: integer, floating point, complex, character, string
  - Binary Number System: Arithmetic, logic circuits, comparison. All data within a computer is defined by bits. Bit: 0/1
- Signals: Acoustic, electrical, electromagnetic, particle, vibration, gravitational forces, temperature

Data Types

- Data types are divided into three basic types: Numeric, Alphabetic, and Alphanumeric. Numeric data consists only of numbers. Alphabetic data consists only of letters and a space character, while alphanumeric data consists of symbols.
- Variables: Numeric – Categorical, Dependent – Independent, Discrete – Continuous
- Binary numbering system, bit: 0/1
- Decimal:
  – Integer (int), Short integer, Long integer
  – Float, Single, Double, Real
- Character
- Boolean
- Arrays
- Constants
- Coefficients

Basic Communication Systems

- A communication system is a setup that transmits information (a message or signal) from one place (the transmitter) to another (the receiver) through some medium (like air, cable, or optical fiber).
- When you speak on a mobile phone, your voice (the message) is converted into electrical or radio signals (transmission), sent over the air (medium), and converted back into sound (reception) at the other end.
- Source → Transmitter → Channel → Receiver → Destination:
  - Voice → converted to electrical signal by the microphone (transmitter).
  - Signal is modulated and sent as radio waves (through the channel).
  - Receiver (other phone) demodulates and converts it back to sound.

Basic Communication Systems

- PSTN (Public Switched Telephone Network- Halka Açık Telefon Şebekesi) şebekesidir. Sabit telefon abonelerini anahtarlayan sistemler.
- GSM (Global System for Mobile- Mobil Haberleşmede Evrensel Sistemi) şebekeleridir.
- Telekom alt yapısı: Devre anahtarlamalı, paket anahtarmalı sistemler kullanılmaktadır.
  - Devre anahtarlamalı sistemler: TDM, PDH, SDH
  - Veri Paket anahtarlamalı sistemler: ATM, Ethernet switch, Router, gateway

Paket anahtarlamalı sistemler:

- LAN (Local Area Network) ve WAN (Wide Area Network):
- LAN: Ethernet Switch,
- WAN: Router/Gateway, Modem.
- İnternet üzerinden veri paylaşımı, ses, görüntü, gerçek zamanlı görüntülü konferans, video gibi hizmetlerini Internet Protokolleri temelinde iletim uygulamaları olarak kabul edilebilir.

Basic Components

A typical communication system has six key parts:

1) Information Source: Produces the message or data to be transmitted. Human voice, text, image, video
2) Transmitter: Converts the message into a suitable signal for transmission. Microphone, modulator, encoder
3) Channel (Medium): The physical path that carries the signal. Air, cable, fiber optic, space
4) Receiver: Converts the received signal back to a usable form. Demodulator, decoder, speaker
5) Destination: The final user of the message. Human ear, computer, display
6) Noise and Interference:
    1) Noise: Unwanted signals that distort the message (e.g., static, interference). Systems use filters, error correction, and digital encoding to reduce noise.

Types of Communication Systems

- Analog Communication:Message is transmitted as continuous signals. AM/FM radio, landline telephone
- Digital Communication: Message is transmitted as binary data (0s and 1s). Internet, mobile phones
- Baseband Communication: Signal is transmitted without modulation. Ethernet cables
- Broadband/Passband Communication: Signal is modulated to a higher frequency for transmission. TV broadcast, satellite link

History of Communication

- In 1831, it was discovered that electricity, first developed, flowed as current through conducting wires. Michael Faraday
- Can messages be sent using electrical signals on conducting wires? Morse and his friend provided the answer in 1840, with the telegraph and Morse Code.
- Can sound be transmitted using electrical signals on conducting wires? In 1876, Scottish-born researcher A. Graham BELL succeeded in transmitting the first human voice over electrical wires in America.
- Can messages be transmitted wirelessly? Tesla (!) and Marconi
- Manual telephone exchanges, relay-switched telephone exchanges: crossbar telephone exchange
- Microprocessor-based computers (in the 1970s)
- Circuit-switched digital exchanges (from the 1980s)
- Multiplexed access systems
- LAN/WAN: Internet, Packet-switched systems
- IoT, smart object communication
- Quantum computing, electrons and photons

Circuit Switching - PSTN

PSTN

History and Evolution of circuit switching

History and Evolution of circuit switching - PSTN

- Manual,
- Strowger,
- Register,
- Cross bar system,
- Trunking,
- Electronic and digital switching systems

PSTN- history

- **Micheal Faraday; 1831, Electric current**
- **Telegraph systems**
  - **1837: Wheatstone and Morse**
- **Telephone**
  - **1876: Alexander Graham Bell**
- **Automatic Switching Exchanges**
  - **1891: Almon Brown Strowger patents first**
  - **1953: C.Clos develops theory of switch architectures**
- **Traffic Theory**
  - **1902: C.E. Molina**
  - **1915: K. D. Erlang**
- **Digitisation**
  - **1939: Alex Reeves invents PCM**
  - **leads to al digital networks (PDH, ISDN)**

PSTN- history

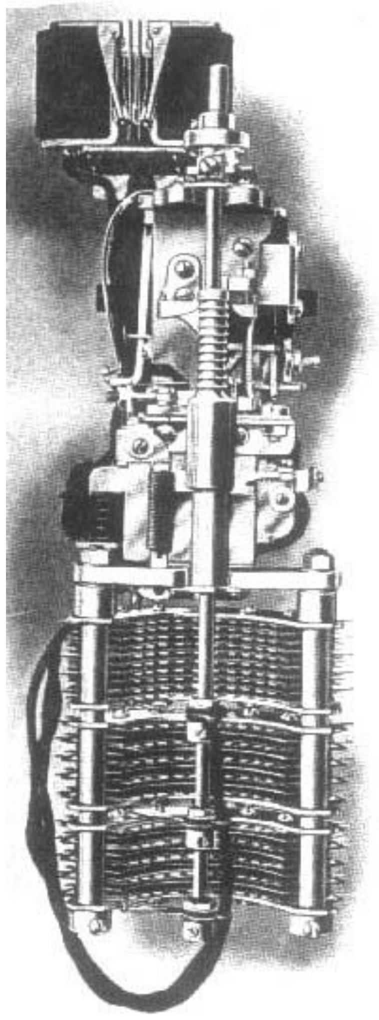
- **Before automatic switching exchange was invented, manual crossbar switchboard was used**

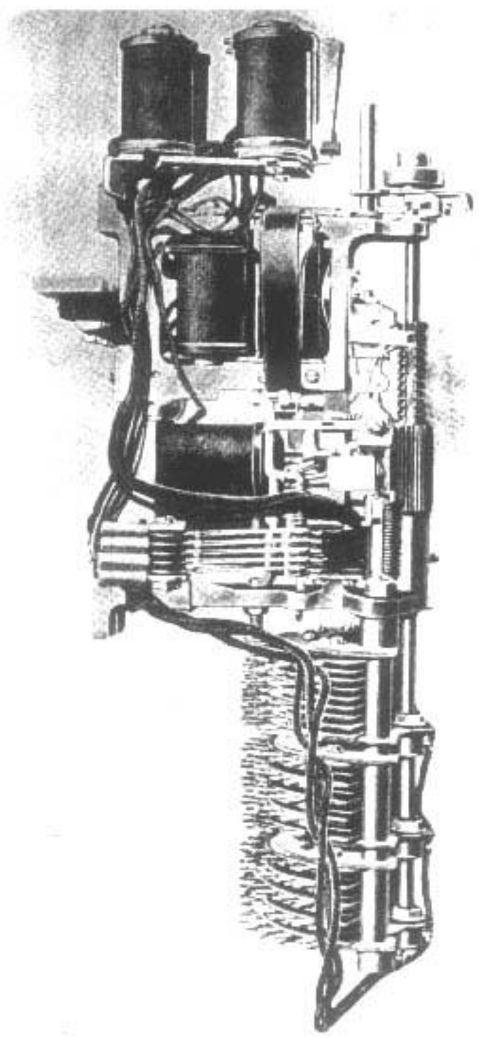
Manual crossbar switchboard

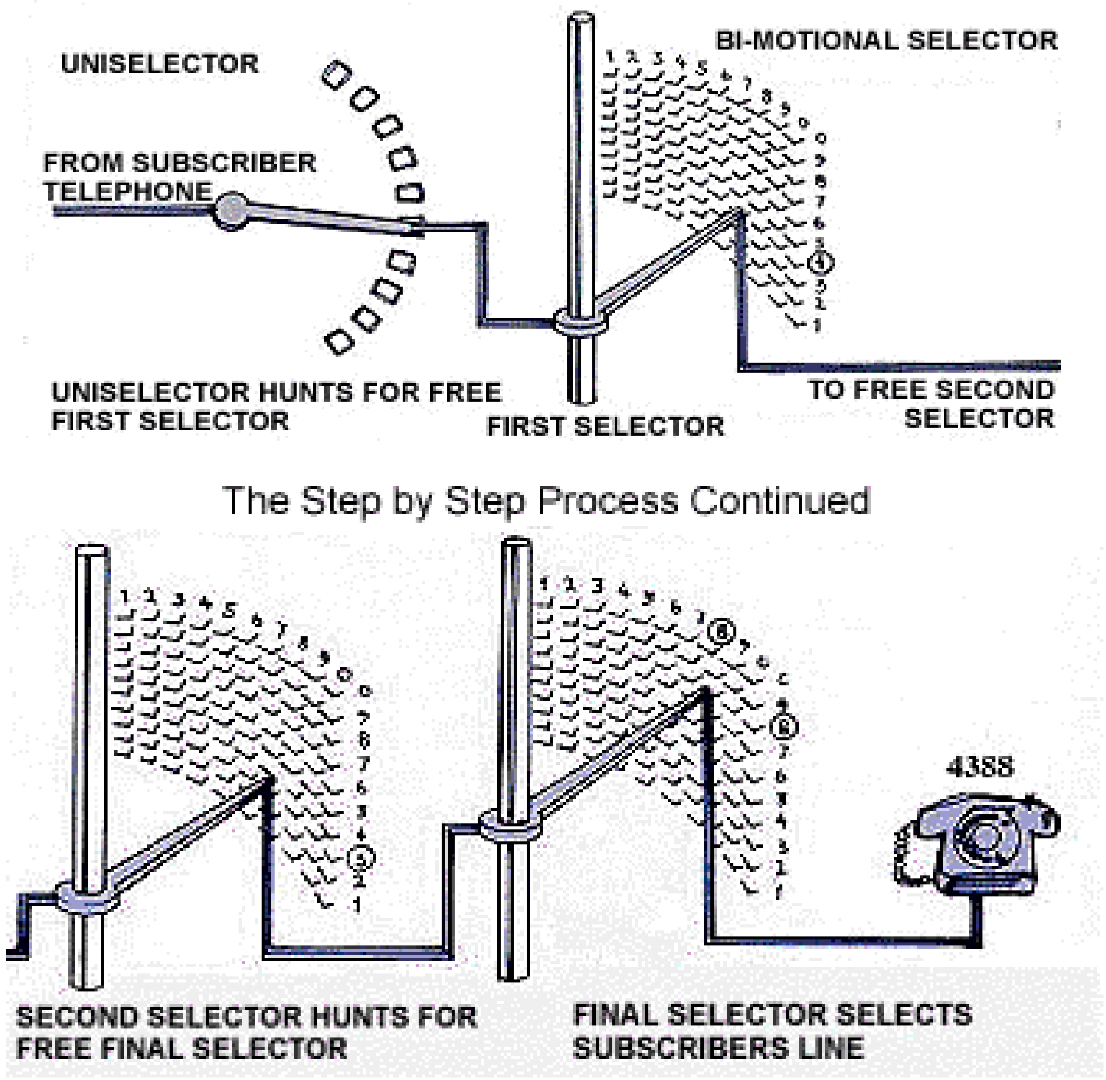
Automatic switching exchange

Guess what was Mr. Strowger's job when he invented the automatic switch?

Strowger step-by-step system







E.g. Dialling no. 4388

PSTN
Public Switched Telephone Network

- The PSTN network has been in use for over a century. When a call is made using this technology, the circuit connection is maintained between the parties until the call is terminated.
- This allows the two parties to establish a circuit, enabling a conversation. Telephone numbers are necessary so that switchboards can determine which cables to connect.
- While connections were initially established manually, the increasing number of subscribers over time necessitated the development of circuit-switched systems.
- Switchboards can be divided into two categories based on their functions: tandem (remote switchboard) and regional.
- A tandem switchboard can connect at least two regional switchboards.
- Consider a Public Switched Telephone Network (PSTN) as a combination of telephone networks used worldwide, including telephone lines, fiber optic cables, switching centers, cellular networks, satellites, and cable systems.

PSTN

- PSTN (Public Switched Telephone Network) stands for Public Switched Telephone Network, or traditional circuit-switched telephone network. It has been in general use since the late 1800s.

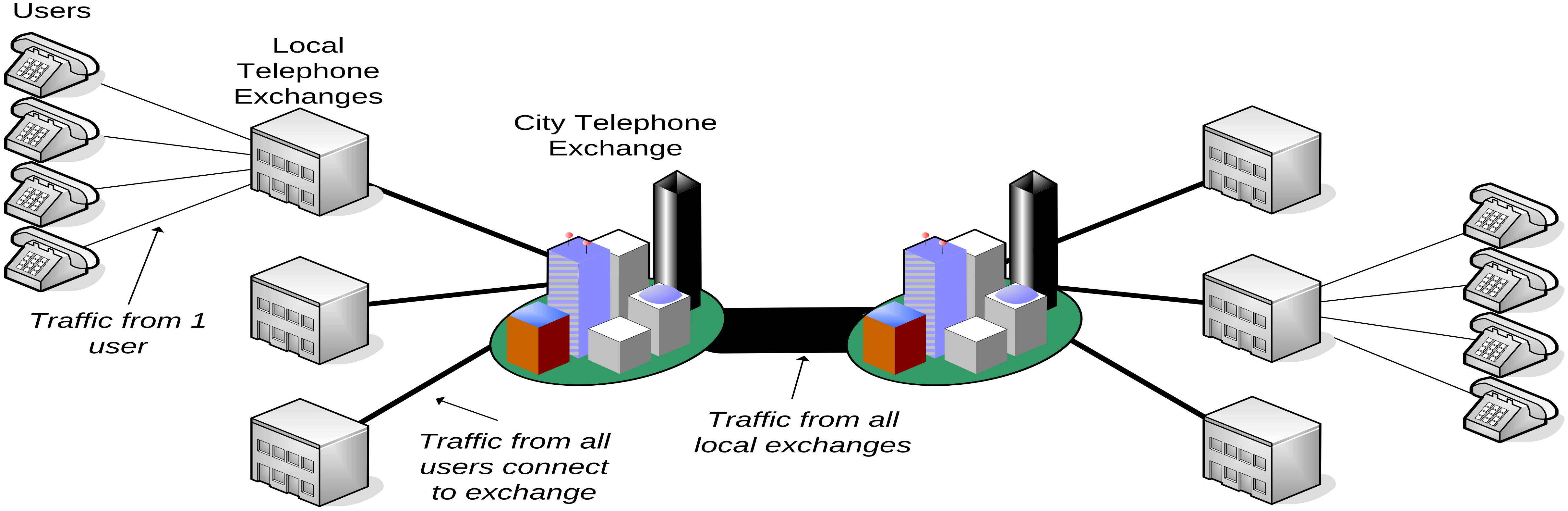
- The PSTN platform, which uses underground copper wires, has provided businesses and households a reliable way to communicate with anyone, anywhere in the world, for generations.

- The telephones themselves are known by various names, including PSTN and landline telephones.

- PSTN telephones are widely used and are still generally accepted as a standard form of communication. However, they have seen a steady decline over the past fifteen years.

- In fact, there are currently only 972 million landline telephone subscriptions in use worldwide, the lowest number yet this century. The number of landlines in Turkey is around 19 million.

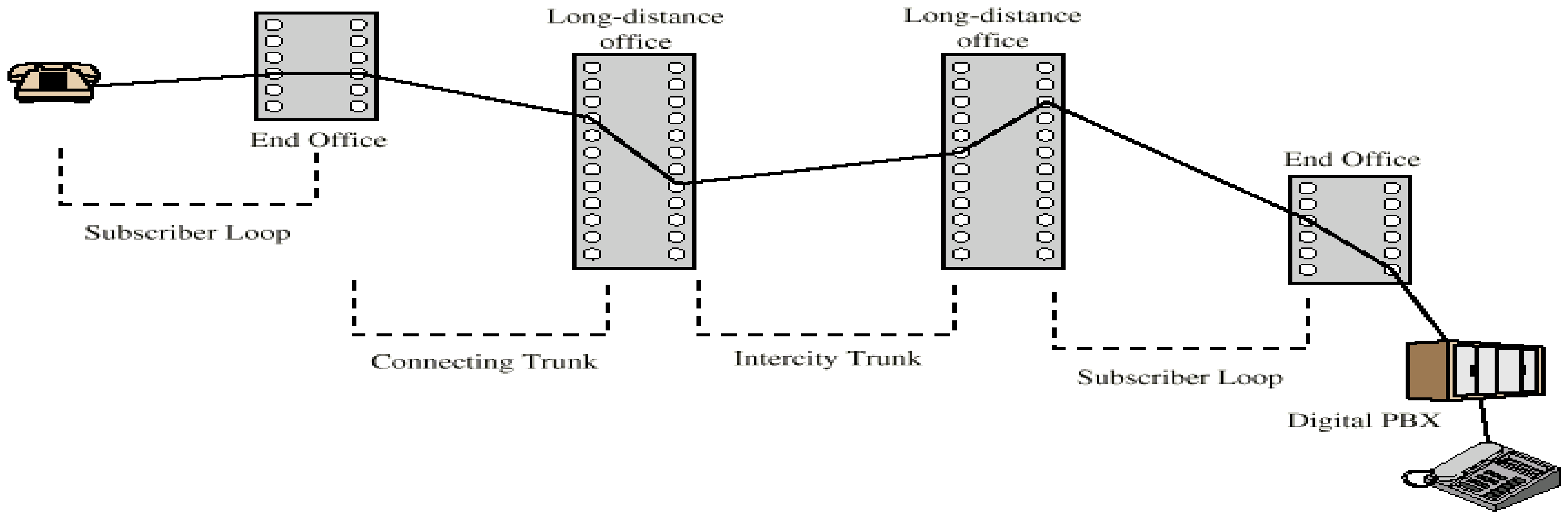
PSTN


Users
Local Telephone Exchanges
City Telephone Exchange
Traffic from 1 user
Traffic from all users connect to exchange
Traffic from all local exchanges

PSTN Working Steps

- Step #1 - Your telephone set converts sound waves into electrical signals. These signals are then transmitted via a cable to a terminal.
- Step #2 - The terminal collects the electrical signals and transmits them to the central office (CO).
- Step #3 - The central office routes calls as electrical signals via fiber optic cable. The fiber optic channel then carries these signals as light pulses to their final destination.
- Step #4 - Your call is routed to a tandem office (a central switchboard responsible for transferring calls to remote central offices) or a central office (for local calls).
- Step #5 - When your call reaches the correct location, the signal is converted back into an electrical signal and then forwarded to a terminal.
- Step #6 - The terminal routes the call to the appropriate telephone number. After receiving the call, the telephone set converts the electrical signals back into sound waves.

Public Circuit Switched Network



Public Switched Telephone Network (PSTN)

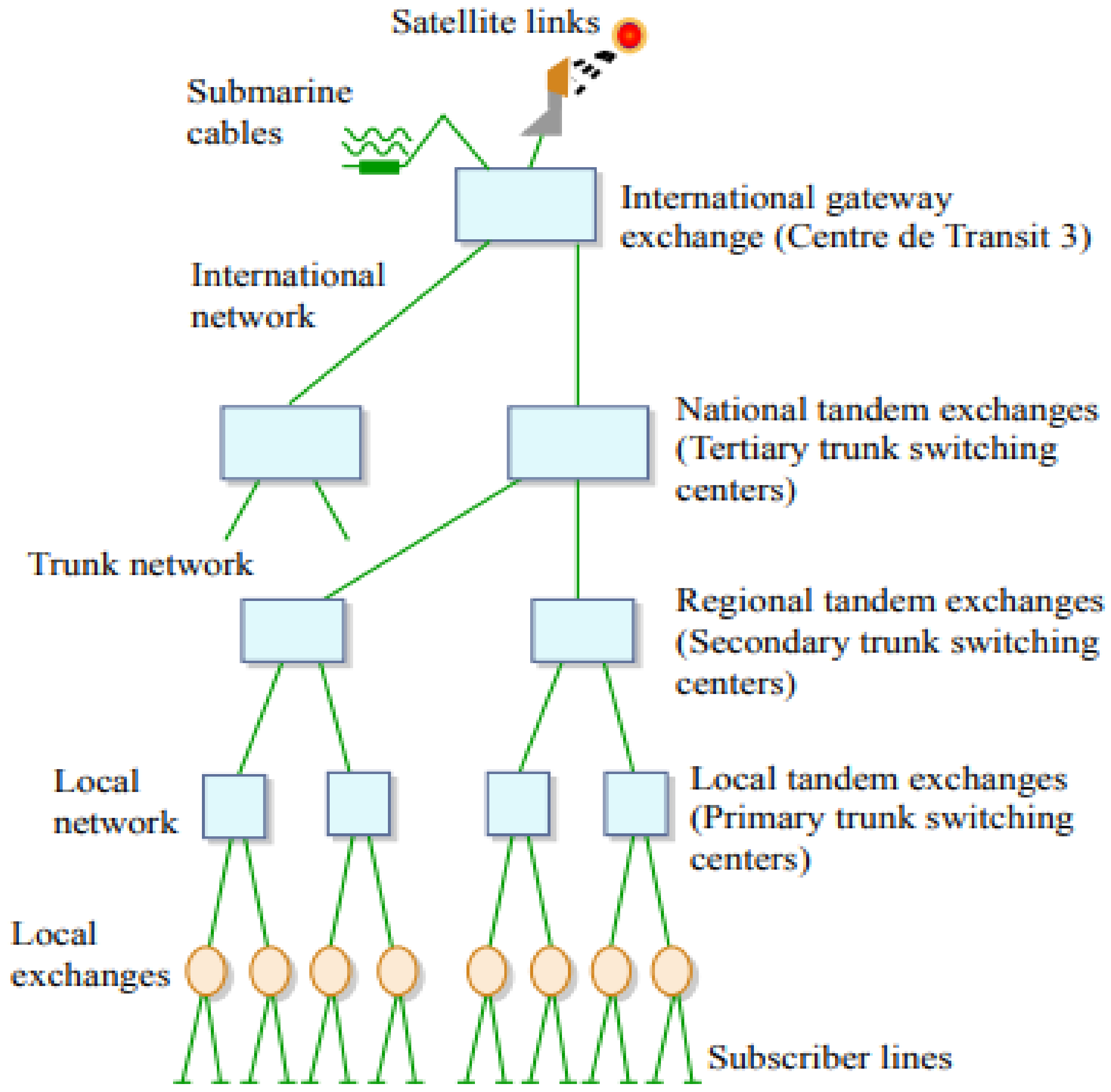
Public Switched Telephone Network (PSTN) consists of switching nodes (called exhanges) in **hierarchical structure**:

a) **Local network** – connect customers' stations to LEs

b) **Junction network** – interconnect group of LEs

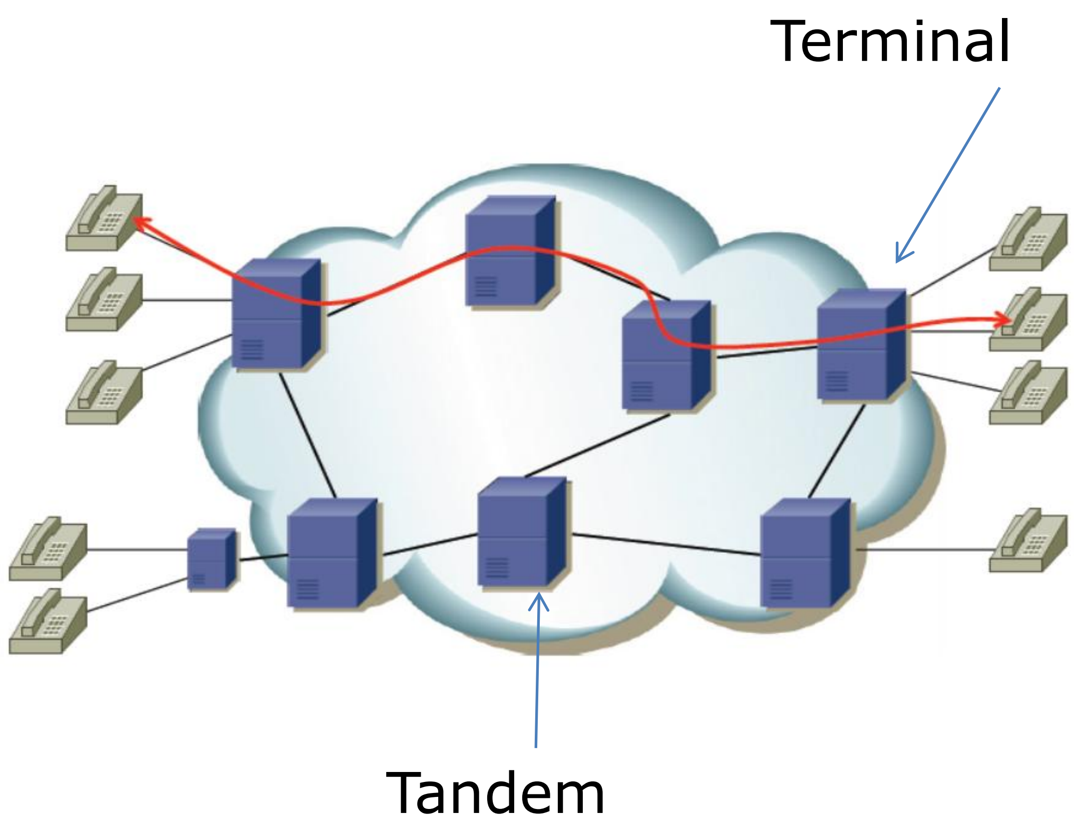
c) **Trunk / toll network** – provides long distance circuits within country

d) **International network** – provide circuits between countries



Terminal – Tandem - Trunk

Terminals: These are the points where telephone lines from subscribers arrive and converge. They are directly switched to each other. They serve from a few hundred to a few hundred thousand subscribers.
Tandem: In PSTN communication systems, the connection between terminals is made via tandem systems.
Trunk: In communication systems, channels dedicated to transmitting signals between two points are called dedicated channels. Typically, these channels are high-bandwidth telephone channels between switching centers capable of handling large numbers of calls and data signaling.



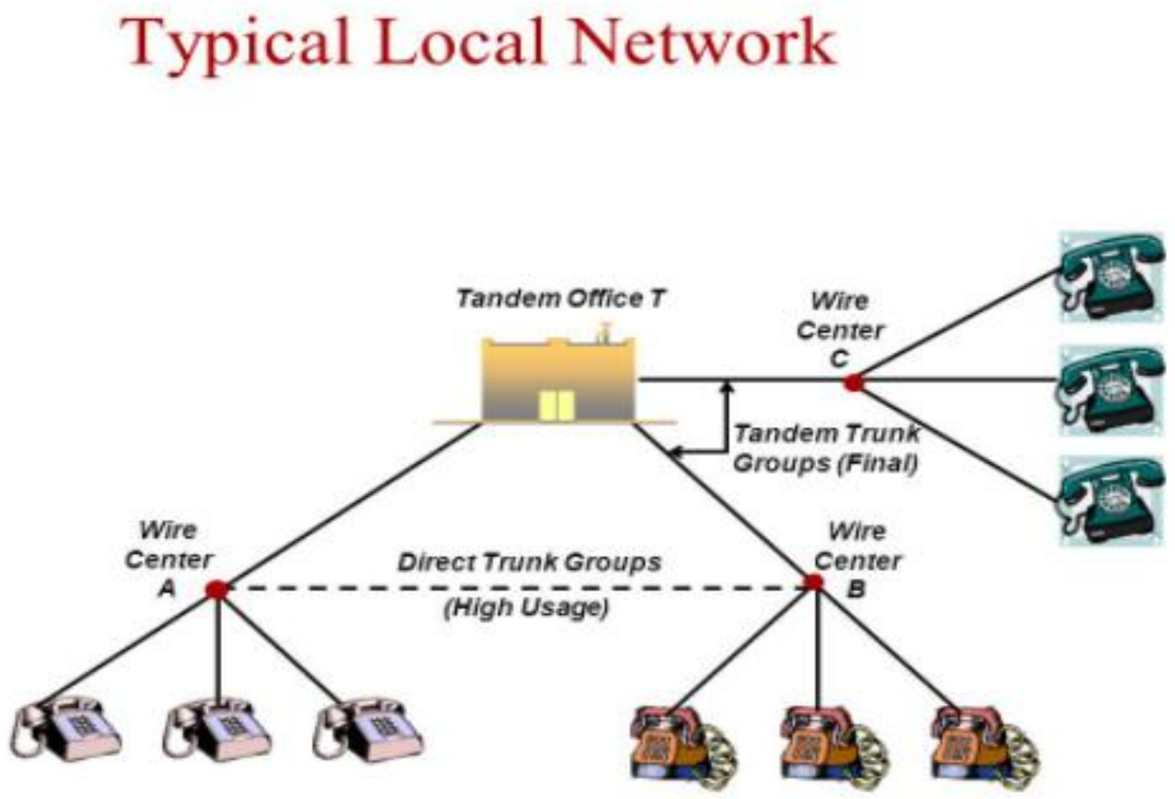
Tandem Circuit Switched (Cascade) Switchboards

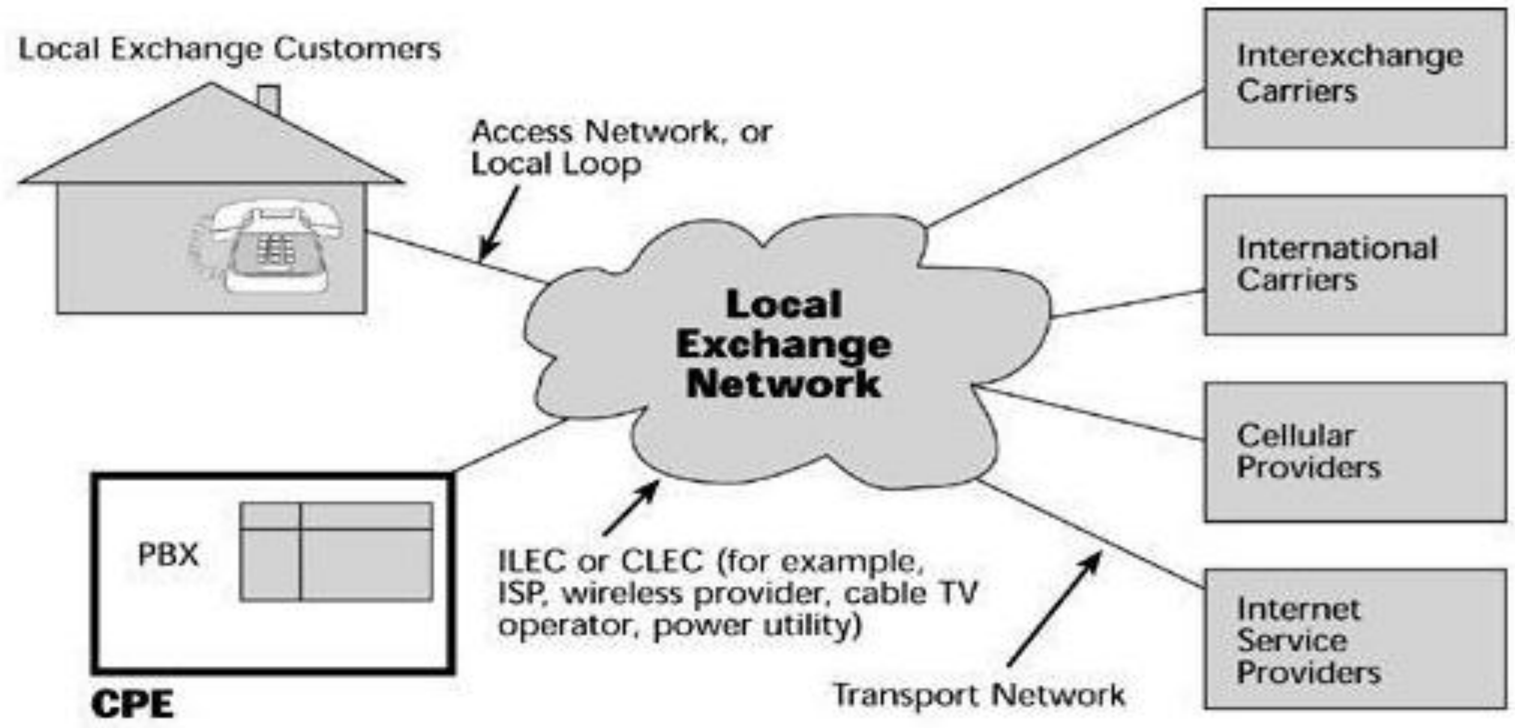
- A tandem office, also known as a network, serves a large geographic area encompassing several local exchanges while managing the switches between them.

- Let's say you call a customer who lives in the same city but in a different suburb.

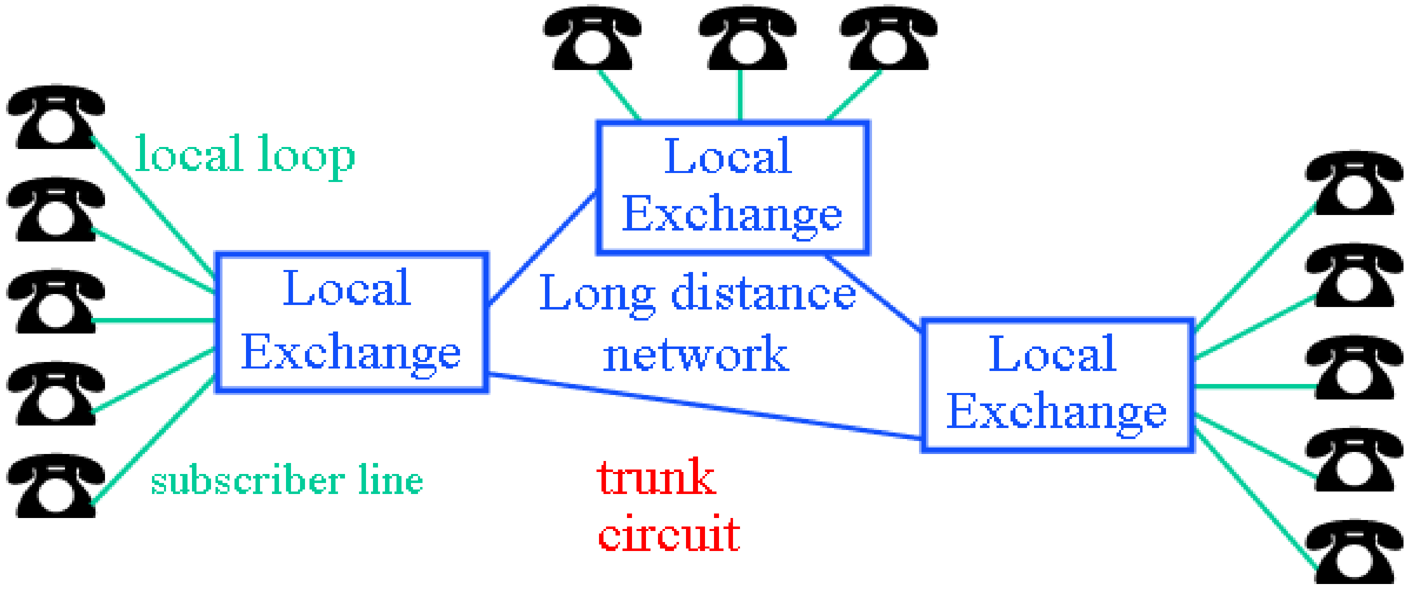
- In this case, your call will be routed from your local exchange to a tandem office, which will then forward the signal to the local exchange near your customer's location.

Local Exchange Network





Local lines are grouped via trunk and long distances are transmitted via tandem switchboards.



The local exchange exchange groups subscribers. The grouped subscribers communicate with each other via trunk circuits. They access long distances via tandem circuits.

PBX - Private Branch Exchange

- Larger organizations with more employees often use dedicated PBX systems.
- A PBX turns your organization into a local switchboard to which all your phone lines are connected, and you enjoy a range of services such as call transfer, call conferencing, auto-attendant, voicemail, call waiting, and more.
- A PBX is connected to your local switchboard. It routes all internal calls through your PBX, while external calls are routed to a local switchboard.
- A PBX is a combination of software and hardware, so it will cost you some serious money. It comes with hubs, switches, phone adapters, routers, and several phone sets. It's like creating your own small switchboard, with your PBX handling the switches internally.
- Most businesses use PBX phone systems to manage calls because they're easier. However, installing and managing a PBX is expensive. The actual cost will vary depending on the number of features you choose and the complexity of the PBX. A huge, complex, and feature-rich PBX will cost your business more than a simple PBX system with only some basic capabilities.

Alternative to PBX : Gateway

- PBXs are used as dedicated switchboards in large organizations. With the advancement of network technologies, the network technology that integrates VoIP and PBX switchboards is the gateway protocol for voice transmission over the internet.

- Switching technology itself hasn't changed much since the last century. Circuit-switched systems are a potential drawback of PSTN telephone networks because they don't allow for the transmission of certain types of data. Packet-switched systems, known in the telephone industry as VoIP, have given rise to a new and modern telephone service.

- VoIP is the transmission of voice and other data over an internet protocol network.
- Your VoIP phone is connected to a DSL or cable modem that connects you to the internet.

- PSTN phones can work with VoIP. You can use an Analog Telephone Adapter (ATA) to convert your traditional phone into a digital phone. It can then be used with your VoIP phone system.

- However, you can't access all the features and add-ons of your VoIP network with these types of landline phones. To access all the features, it's best to use a SIP-enabled phone. After all, it's all about features, so what's the point of having a VoIP network if you don't want to use any additional features?

Switching in Digital Switchboards

PCM: Pulse Code Modulation

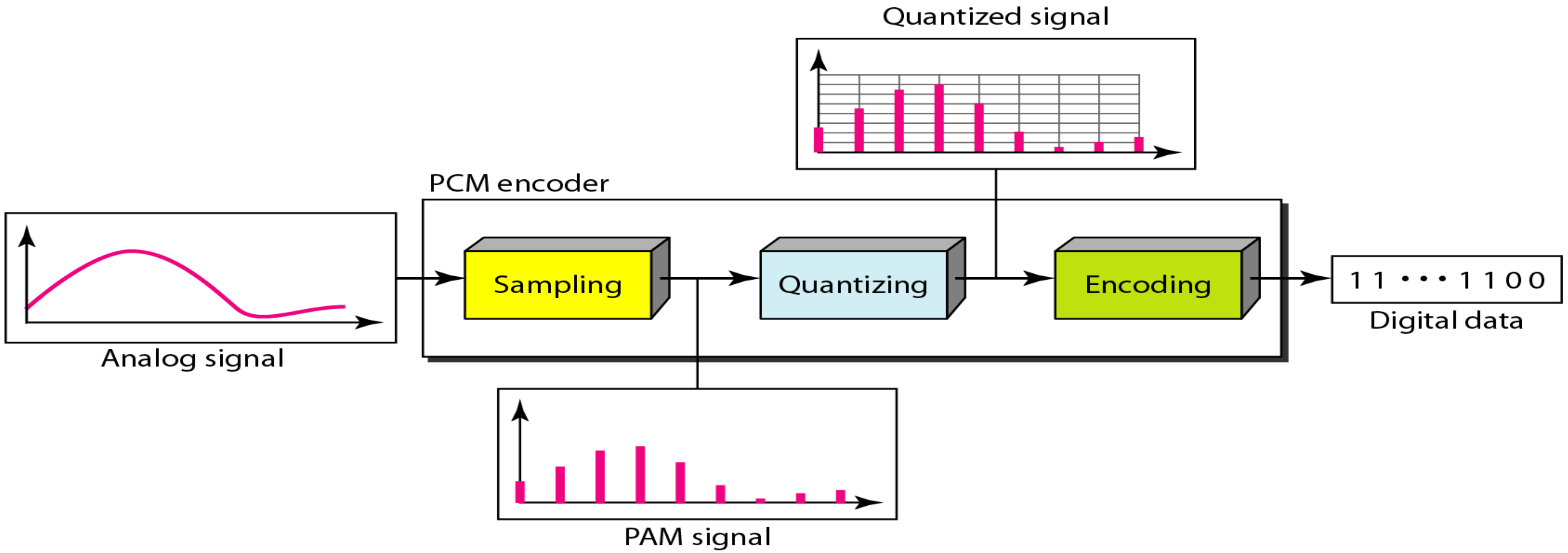
Why is digital switching needed?

- Sound is an acoustic signal; for transmission, it must be converted to an electrical signal (microphone).
- A speaker is used to convert the electrical signal back to an acoustic signal.
- An electrical signal is an analog signal. For digital switching and transmission, the analog signal must be converted to a digital signal (bit: 0/1). An analog signal consists of a mixture of many sinusoidal signals.
- The basic name for the system that performs this function is PCM.

PCM

- The PCM (Pulse Code Modulation) method involves three steps to digitize an analog signal:
  - Sampling: The process of taking samples of amplitude values at fixed time intervals. This produces discrete values.
  - Quantization: The process of scaling the discrete values obtained through sampling to specified amplitude values. Errors are inherent.
  - Binary encoding: The conversion of discrete values obtained through quantization into the binary number system. Symbol: The binary number that represents the number of intervals. For example, if bit length = 8 bits, the number of symbols = 2^8 = 256.
- Before sampling, we need to filter the signal to limit the maximum frequency of the signal, as this affects the sampling rate. (Speech signals between 0Hz and 4000Hz are allowed to pass.)
- The International Standards Organization has set the speech bandwidth at 4kHz because it provides these characteristics at a minimum level. The basic criteria that speech must meet in this determination are: recognition, understanding, perceptibility, and latency.
- Filtering should ensure that we do not distort the signal and that high frequency components that negatively affect the signal shape are removed.

Analog sinyalin sayısallaştırılmasında Quantalama



Örnekleme alındığında herbir ayrık değerin rakamsal karşılığı vardır. Fakat ikili sayı sisteminde belirli bir bit sayısı ile temsil edilen ayrık değer ile örnekleme ile elde edilen ayrık değer farklı olduğunda quantalama hatası ortaya çıkar.

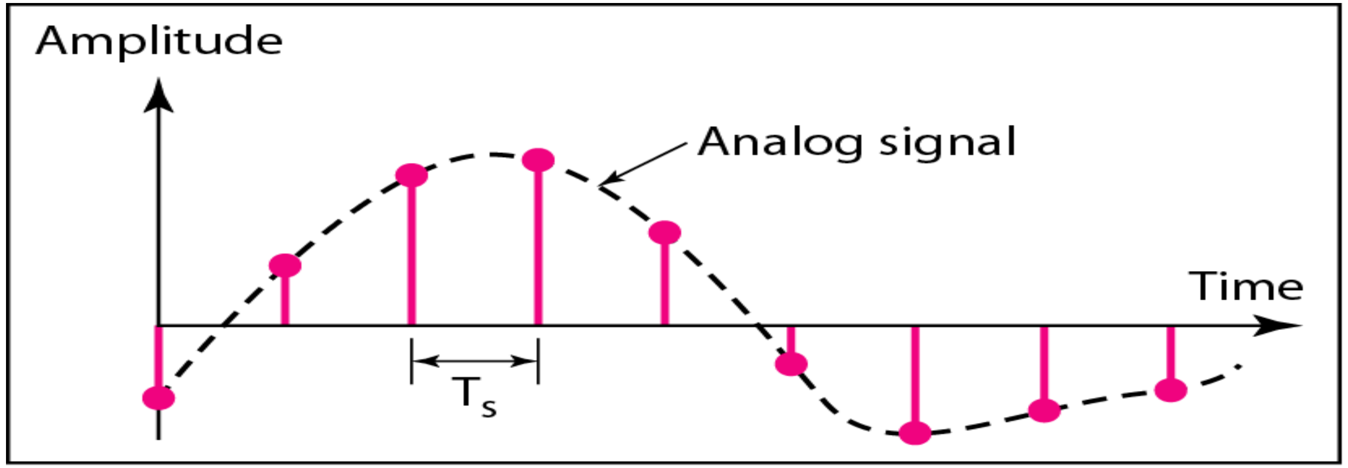
Pulse Code Modulation Standarts

- Communication systems are manufactured according to international standards. Standards: CCITT, AT&T, ...
- PCM is a Time-Domain Waveform coding method and is defined within CCITT G.711, and AT&T 43801.
- Basically, an analog signal is sampled at a frequency of 8000 Hz. In each sample, the amplitude of the signal is assigned (quantized) a digital value (8 bits).
- Nyquist theorem: When an analog signal is sampled, converted to digital, transmitted, and then converted back to analog, to achieve the same quality as the original signal, the sampling frequency must be equal to or greater than twice the bandwidth or maximum frequency. (Sampling interval = 1/sampling frequency). (For telephone communication, the sampling frequency is taken to be twice the sampling frequency; if you're listening to music or developing a home theater application, it's taken to be greater than twice the sampling frequency.
- In telecom, the sampling interval is T=1/fs; fs: sampling frequency. T=1/8KHZ=1/8000=125 microseconds.
- The International Standards Organization has set the bandwidth as 4KHz. Why?
- There are two PCM algorithms defined within CCITT G.711, called "A-Law" and "Mu-Law." Mu-Law PCM is used in North America and Japan, and A-Law is used in most other countries. In both A-Law and Mu-Law PCM, the values used to represent the amplitude are a number between 0 and +/- 127; therefore, 8 bits are required to represent each sample (2 to the eighth power = 256).
- It can be seen then that PCM operates at a rate of: 8 bits/sample * 8000 samples/sec = 64000 (64K) Bits Per Second. If an analog signal is sampled every 125 microseconds and transmitted as 8 bits, how many bits are transmitted in 1 second?

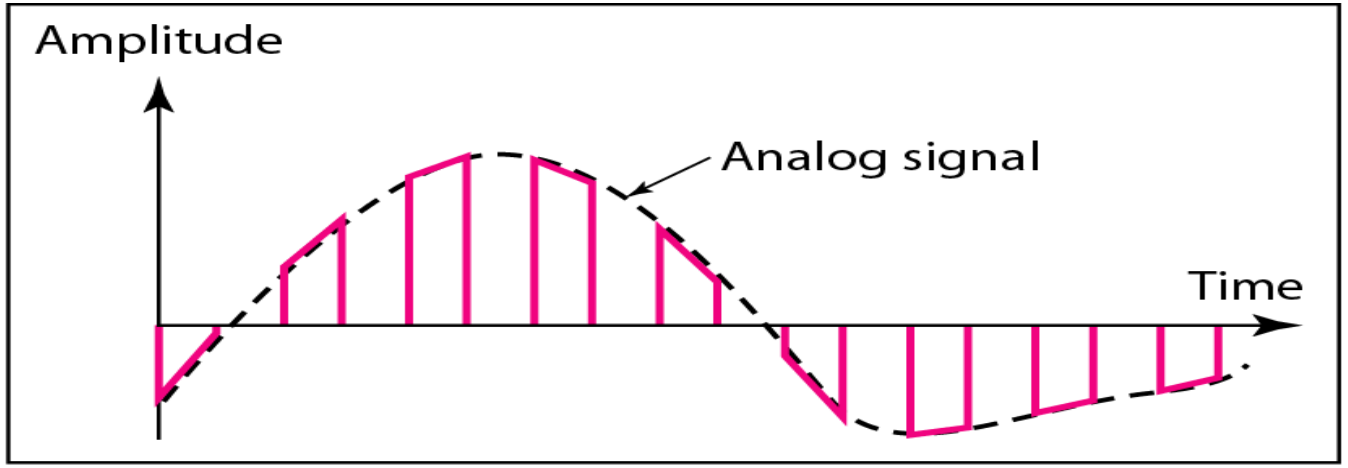
Conversion of speech signal into digital signal

- The human voice signal propagates as a mechanical pressure wave in the air.
- The human voice is converted into an electrical signal. This signal is an analog signal. An analog signal is a signal whose amplitude, frequency, and phase vary over time.
- The human hearing frequency range is between 20 Hz and 20,000 Hz. This range decreases with age.
- In communication systems, the speech and hearing frequency range is taken as 0 Hz to 4,000 Hz. Bandwidth, BW = 4,000 Hz = 4 KHz
- In communication systems, sound is converted into an analog signal. The bandwidth of this analog signal is 4 KHz.
- If the bandwidth of an analog signal, BW = 4 KHz, the sampling frequency is fs = 8 KHz. Because, fs ≥ $2*BW$
- If the sampling frequency of an analog signal is 8 KHz, how many Hz is it in? If the sampling interval, Ts = 1/fs, then Ts = ? Note, fs is taken as Hz. Ts=1/8000=125 microseconds.
- If a sample is taken from the analog signal of a human voice every 125 microseconds, and this sample is represented and transferred with 8 bits, 64,000 bits are transferred in 1 second.

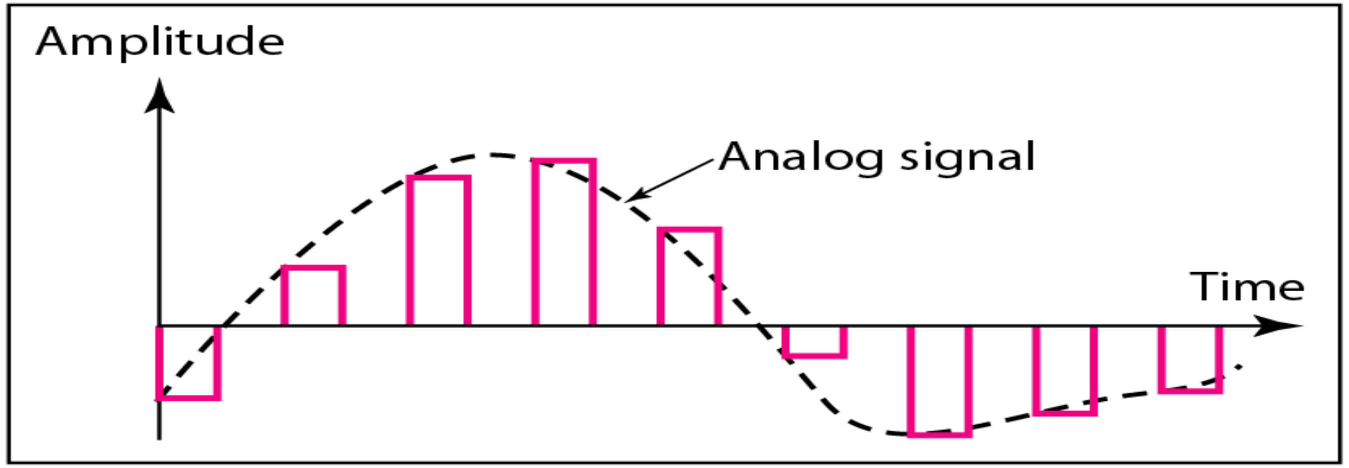
Three different sampling methods for PCM



a. Ideal sampling



b. Natural sampling



c. Flat-top sampling

Example

We want to digitize (ayrık) the human voice. What is the bit rate, assuming 8 bits per sample?

Solution

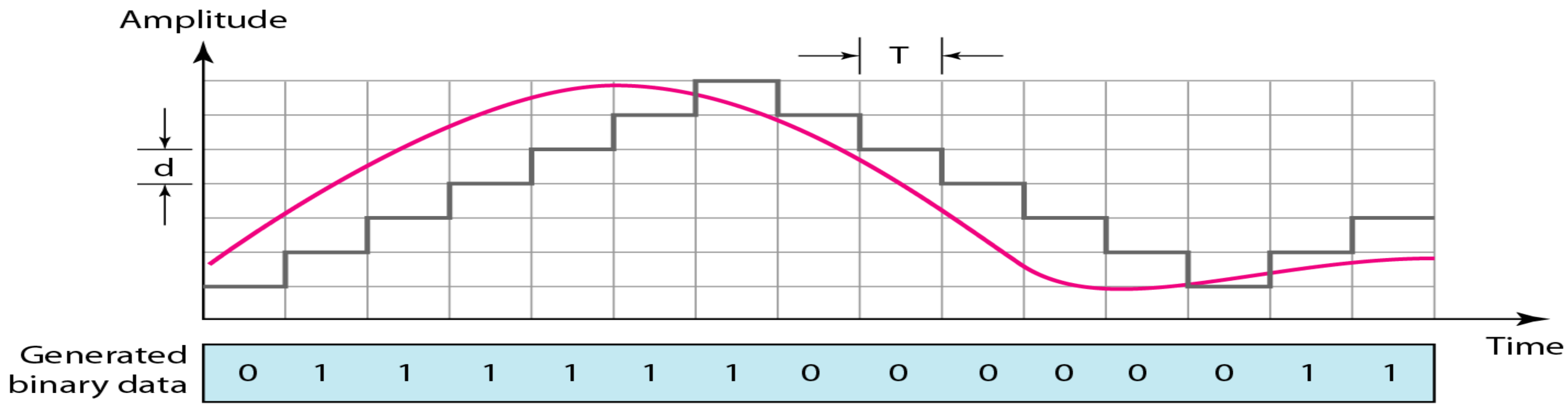
The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:

$$\text{Sampling rate} = 4000 \times 2 = 8000 \text{ samples/s}$$
$$\text{Bit rate} = 8000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

Delta Modulation

- This scheme sends only the difference between pulses, if the pulse at time t_{n+1} is higher in amplitude value than the pulse at time t_n, then a single bit, say a "1", is used to indicate the positive value.
- If the pulse is lower in value, resulting in a negative value, a "0" is used.
- This scheme works well for small changes in signal values between samples.
- If changes in amplitude are large, this will result in large errors.
- Instead of using one bit to indicate positive and negative differences, we can use more bits -> quantization of the difference.
- Each bit code is used to represent the value of the difference.
- The more bits the more levels -> the higher the accuracy.

The process of delta modulation

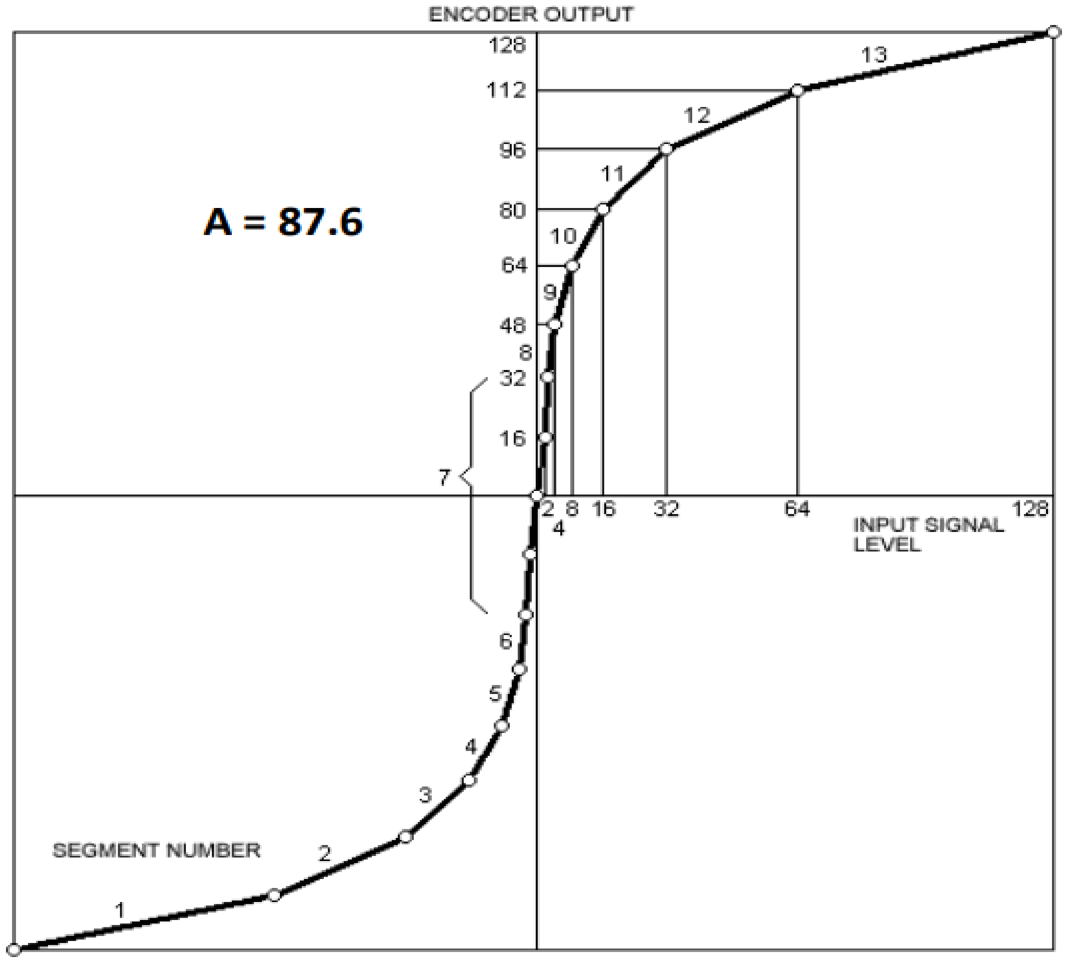


Speech Compression Standards

- 64 kbps μ-law/A-law PCM(CCTT G.711)
- 32 kbps ADPCM(CCITT G.721)
- 16 kbps Low Delay CELP(CCITT G.728)
- 13.2 kbps RPE-LTP(GSM 06.10)
- 13 kbps ACELP(GSM 06.60)
- 13 kbps QCELP(US CDMA Cellular)
- 8 kbps QCELP(US CDMA Cellular)
- 8 kbps VSELP(US TDMA Cellular)
- 8 kbps CS-ACELP(ITU G.729)
- 6.7 kbps VSELP(Japan Digital Cellular)
- 6.4 kbps IMBE(Immarsat Voice Coding Standard)
- 5.3 & 6.4 kbps True Speech Coder(ITU G.723)
- 4.8 kbps CELP(Fed. Standard 1016-STU-3)
- 2.4 kbps LPC(Fed. Standard 1015 LPC-10E)

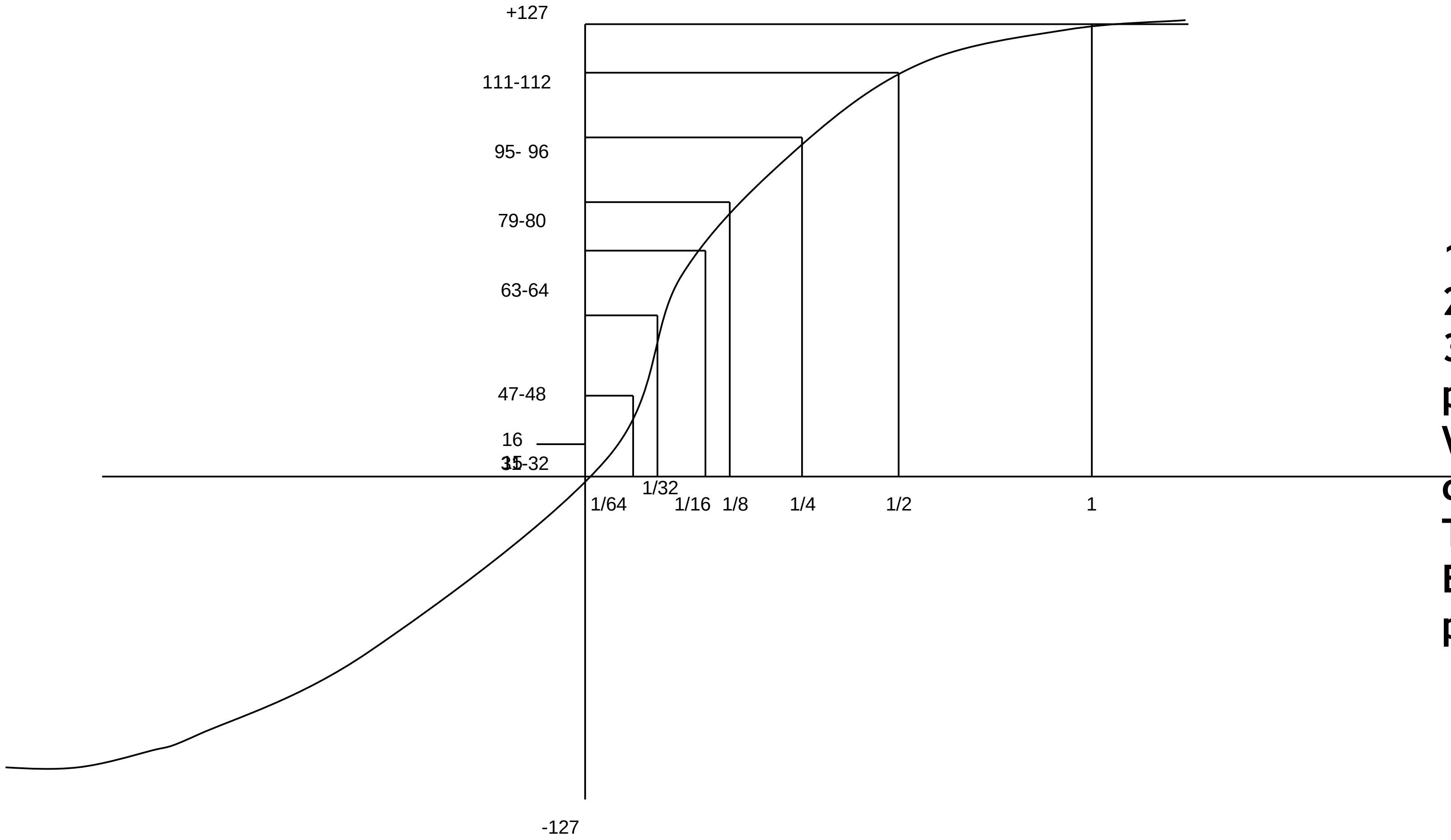
A-Law – Quantification standard

A-law Compandor (Europe) Shown Graphically



- In communication systems, analog signals cannot be sampled linearly and with equal amplitude intervals at equal time intervals.
- Non-linear sampling is performed at equal time intervals. The A-law standard on the right contains the non-linear sampling intervals applicable to our country. For small amplitudes, more frequent amplitude levels are considered.
- Why?
- When small amplitude speech is lost, a fundamental communication principle, quality problems, arise. Slow speech is crucial for privacy.

QUANTALAMA TEKNİĞİ NON-LİNEER QUANTALAMA



2^8 = We can encode 256 signs.

1. The first bit indicates the positive or negative region.
2. Bits 2, 3, and 4 indicate which part it is in. (010)
3. Bits 5, 6, 7, and 8 indicate where it is located within the part.

We can create 16 different codes with these four characters.
There are 15 codes in each part where it appears.
EXAMPLE: How is the 45 encoded in the positive second part? 1 0 1 0 1 1 0 1

In a linear quantization system, low-amplitude signals have a high error rate. A large portion of the voice of someone speaking quietly on the phone is lost. To prevent this, we divide the amplitude axis into equal parts. Low-amplitude signals have tighter intervals, while high-amplitude signals have fewer intervals. We adjust the intervals so that the signal-to-noise ratio remains constant. This process is called non-linear quantization.

TDM - E1
Time Division Multiplexing

Time Division Multiplexing

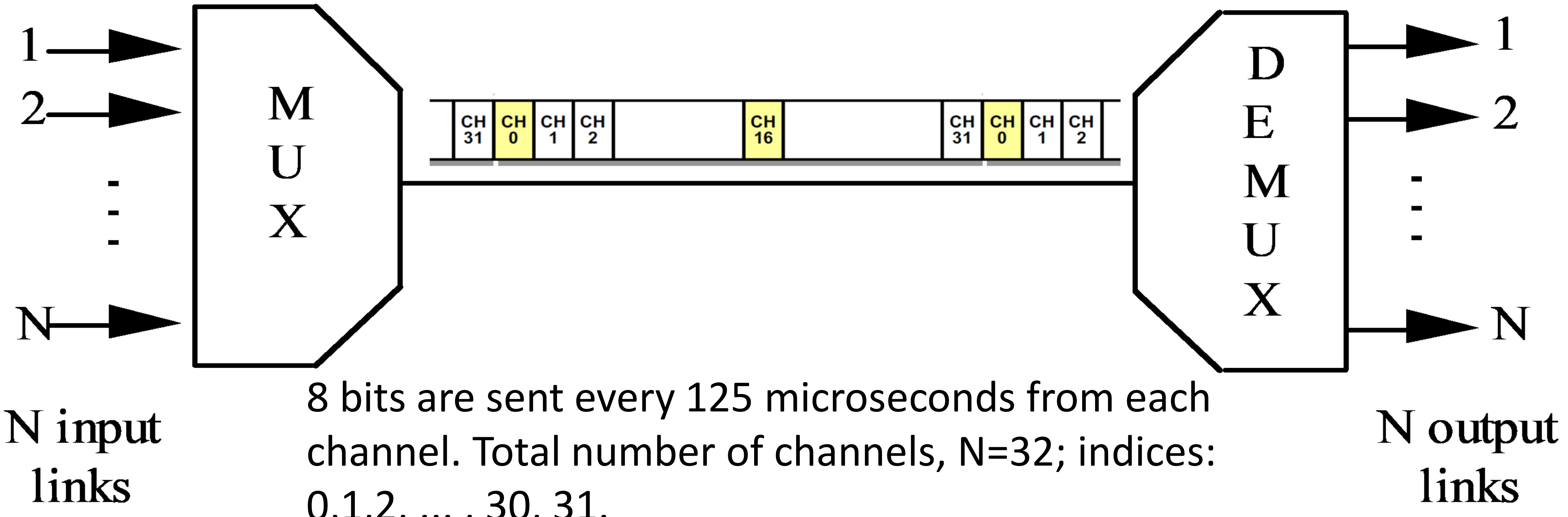
- Time Division Multiplexing (TDM) is a data transmission technique that allows multiple signals to share the same communication channel by dividing the available time into separate time slots. Each signal is assigned a specific time slot during which it can transmit its data.
- It answers the question of how to communicate with many subscribers simultaneously in a communication environment, instead of making one phone call from a single line.
- The total transmission time on a communication channel is divided into equal time intervals (time slots).Each user or data source gets its own time slot in a repeating sequence.The signals are sent one after another, not simultaneously, but so fast that they appear continuous.
- Suppose there are 4 users (A, B, C, D) sharing one channel. The system divides each second into 4 time slots: one for each user. During slot 1, A sends data; during slot 2, B sends data; and so on. After D finishes, the cycle repeats continuously.
- **Applications:**
  - Telephone systems (digital telephony)
  - Satellite and optical fiber communications
  - Computer networks and data links
  - Multiplexed data acquisition systems

Sampling in Communication Systems

- Sound waves are acoustic waves, which travel through air under pressure. Sound waves are converted into electrical signals. The bandwidth of this signal is between 0Hz and 16000Hz. In telephone communications, the bandwidth is taken as B=4KHz. This bandwidth corresponds to the understanding, recognition, and sensation properties of sound.
- The Nyquist formula is used to calculate noise-free channel capacity. C=2Bm, bits/second; where B is the bandwidth Hz; m=8 is the number of bits representing the symbol; the number of bits corresponding to the sample received from the analog signal. C=2*4000*8=64000bit/second=64Kbit/sec.
- The Shannon capacity formula is used to calculate noisy channel capacity. C=B log2(1+SNR), SNR: The signal-to-noise ratio. SNR=2(C/B)-1. SNR=2^(64000/4000)-1=2^16-1. An approximate SNR=2^16. SNRdB=10log10(2^16)=160*0.3=48dB. (Log10(2)=0.3)
- The Nyquist Sampling Theorem states that fs≥2*B or fs≥2*fmax. In telephone communications, fs=2*B is used. fs=2*4KHz=8KHz. The sampling interval is Ts=1/fs=125 microseconds.
- If an 8-bit sample is taken from a voice signal every 125 microseconds and transmitted, 64000 bits/sec are sent in one second.

TDM – E1 (2048Kbps)

- Time division multiplexing allows a link to be utilized simultaneously by many users. N=32



8 bits are sent every 125 microseconds from each channel. Total number of channels, N=32; indices: 0,1,2, ... , 30, 31.

What is E1?

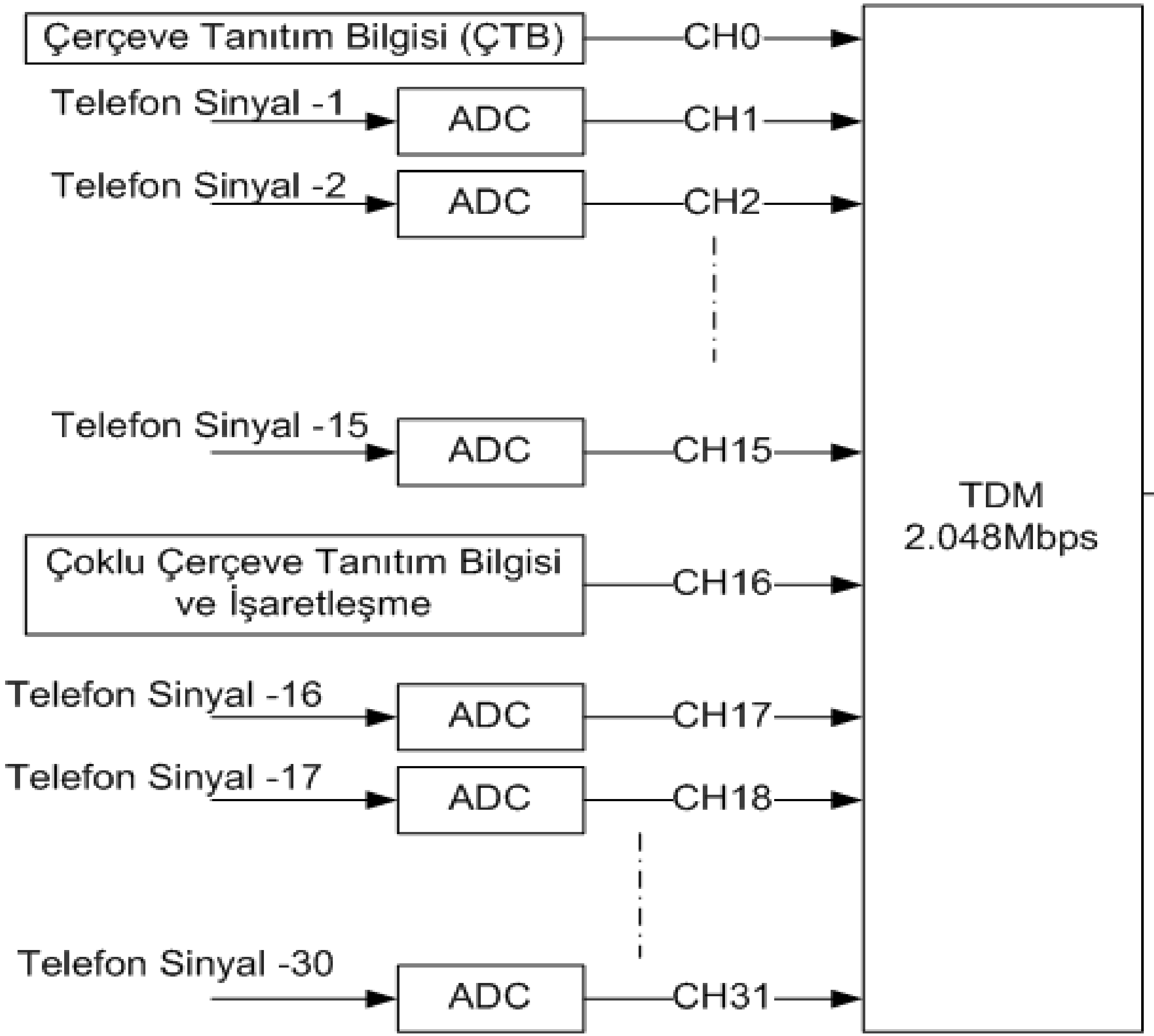
- E1 is a digital transmission standard that uses TDM to carry multiple voice or data channels over a single physical link.It is mainly used in Europe and Asia, while T1 is its counterpart in North America.
- Basic Characteristics of an E1 Line:
  - Total Bit Rate 2.048 Mbps
  - Number of Channels 32 time slots
  - Speed per Channel 64 Kbps
  - Multiplexing Technique TDM (Time Division Multiplexing)
  - Frame Duration 125 microseconds
  - Standard ITU-T G.703 / G.704

How E1 Works

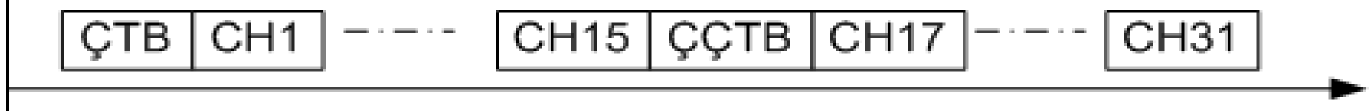
- The E1 frame is divided into 32 time slots (TS0–TS31):
  - TS0: Frame synchronization and control.
  - TS1–TS15, TS17–TS31: Carry user data (voice or digital information).
  - TS16: Signaling (call setup, line control, etc.).
  - Each frame repeats every 125 microseconds, meaning 8000 frames per second are transmitted.
  - This allows 30 voice channels plus signaling and synchronization to share a single 2.048 Mbps link.
- Think of an E1 line as a 32-seat bus:
  - Each seat represents a time slot (channel).
  - The bus (the communication line) makes a trip every 125 microseconds.
  - Each passenger (signal) gets their turn to “talk” during their time slot.
  - Even though only one speaks at a time, everyone’s message gets through — that’s the power of TDM.

TDM – E1: 2.048Mbps

uzaktaki nokta kanalların konumları nasıl tanımlayacak

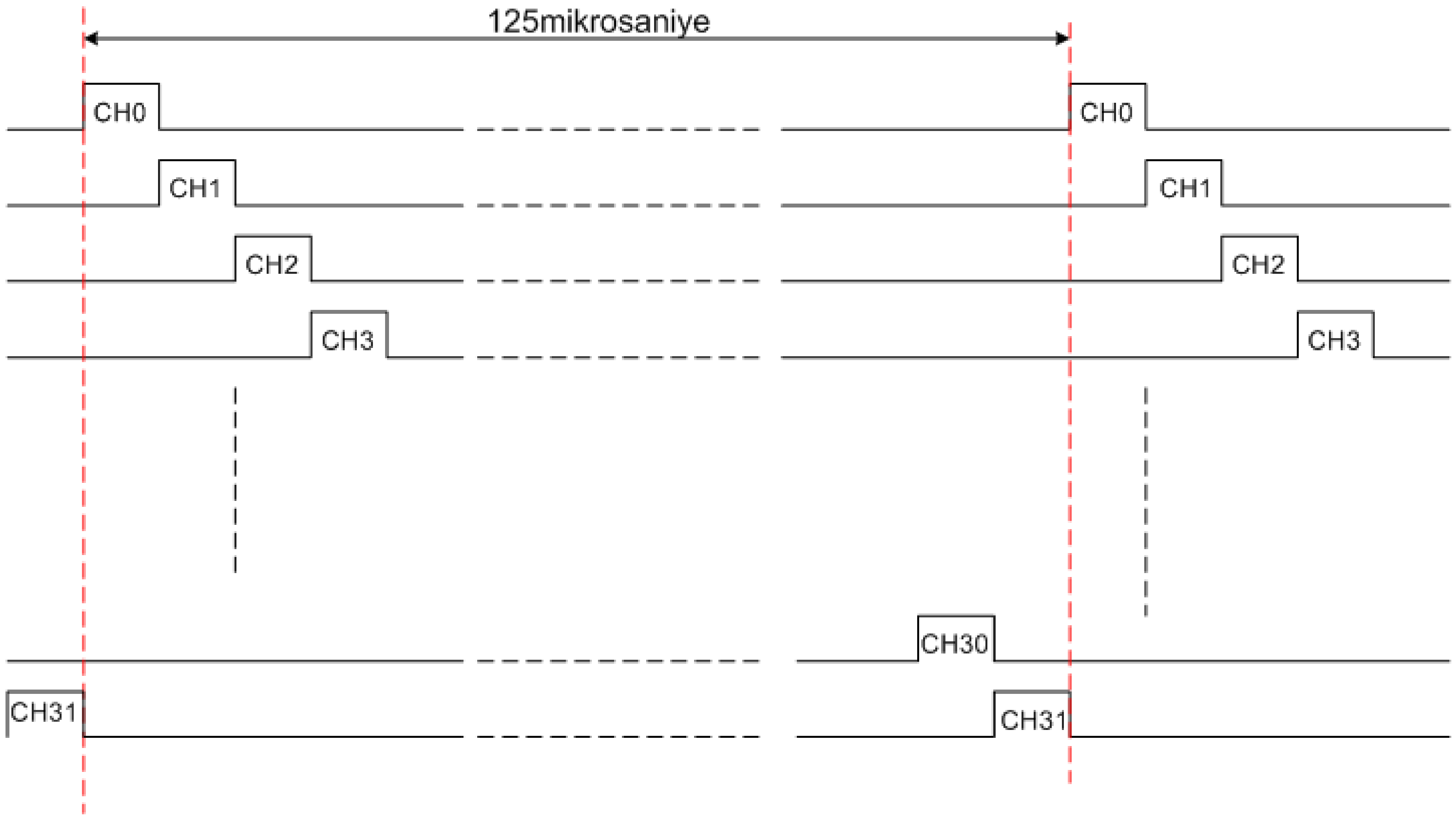


- Devre Anahtarlamalıdır
- 125mikrosaniyede 32 kanaldan 32*8=256bit gönderilir.
- Bir çerçeve 32 kanaldan oluşur.



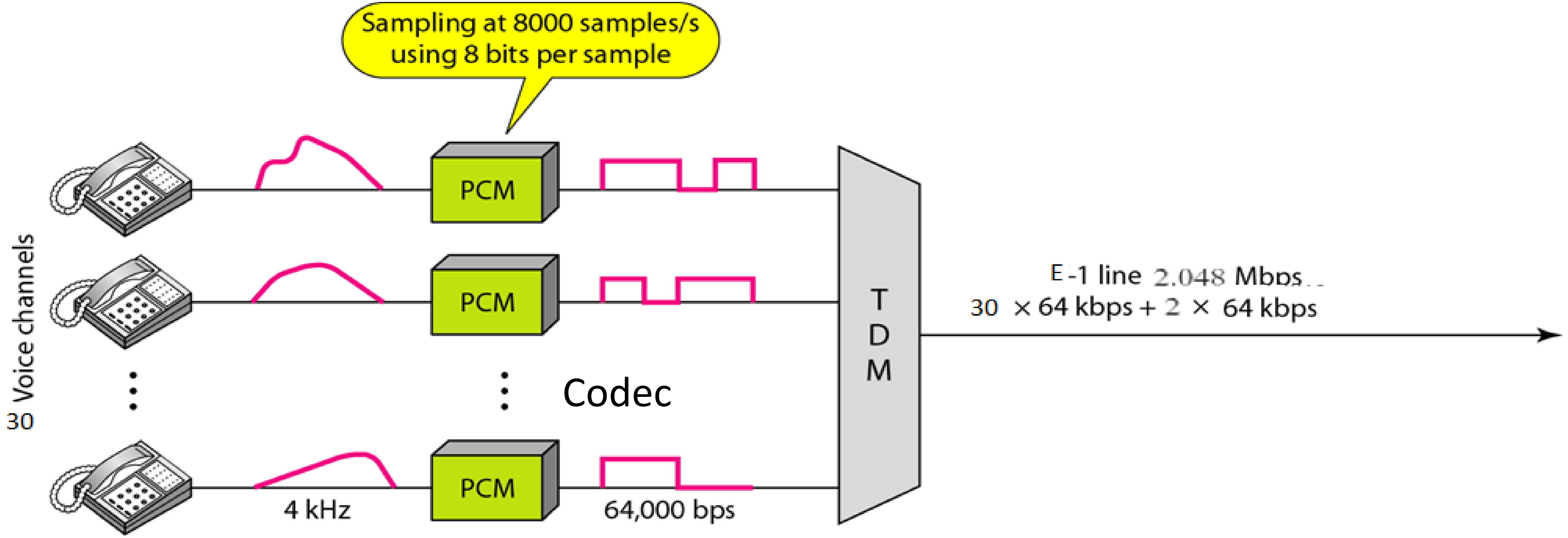
- 125 mikrosaniyede bir çerçeveden 256 bit transfer edilirse, 1 saniyede bir çerçeveden kaç bit transfer edilir.
- X=256/125mikrosaniye=256/(125*10^-6) = 2.048Mbit/sec

TDM – E1: 2.048Mbps - Clock



8 bits are sent from each channel.

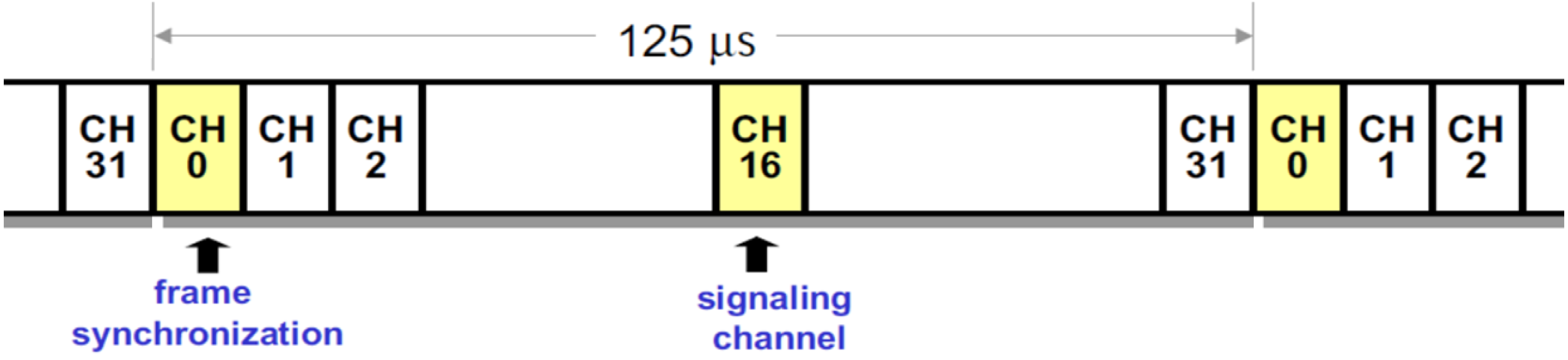
E-1 line for multiplexing telephone lines



- There are 32 channels in the E1 TDM multiplexing circuit. 30 of these channels are communication channels. The reverse channel and two speech channels are used to transfer frame identification information and signaling information.
- There are 32 channels in the E1 TDM multiplexing circuit. If 64,000 bits are transferred per second from one channel in an E1 circuit, then 2,048,000 bits are transferred per second across the entire E1 circuit.

E1-frame

30 data channels+2 control channels



E1 bit rate : 32x 64Kbps = 2.048 Mbps

PCM COMMUNICATION SYSTEMS

The plesiochronous digital hierarchy (PDH) has two primary communication systems as its foundation. These are the T1 system based on 1544kbit/s that is recommended by ANSI and the E1 system based on 2048kbit/s that is recommended by ITU-T.

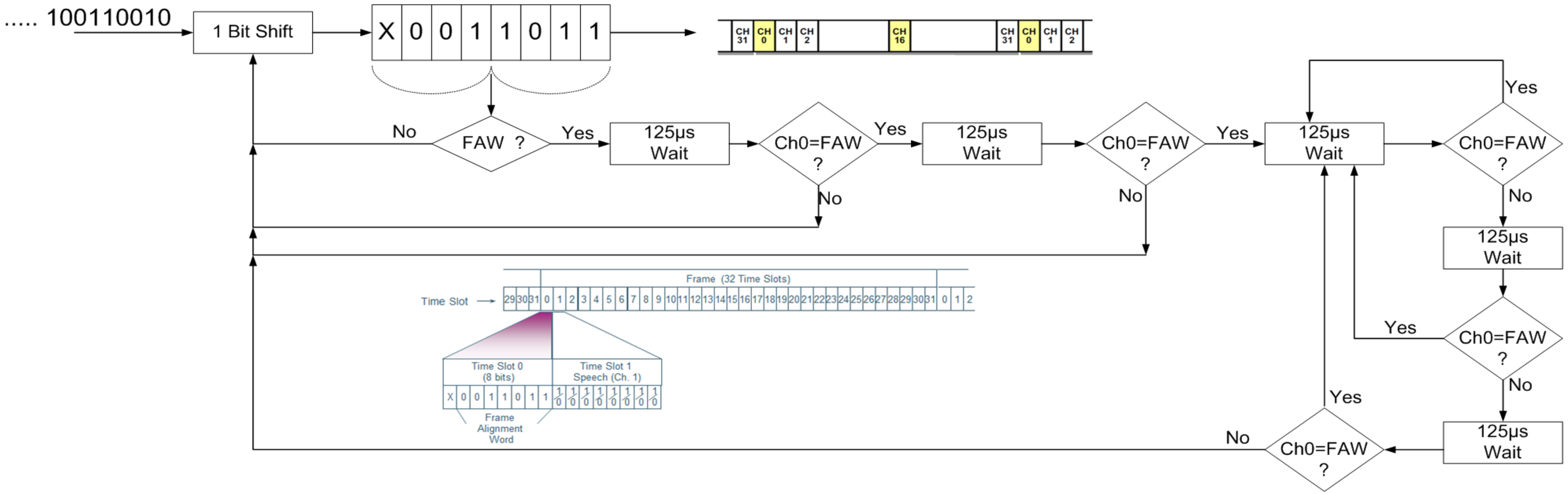
The T1 system is used mainly in the USA, Canada and Japan. European and certain non-European countries use the E1 system.

	Common characteristics	
a	Sampling frequency	8kHz
b	Number of samples per telephone signal	8000 per second
c	Length of PCM frame	1/b = 1/8000/s = 125µs
d	Number of bits in each code word	8
e	Telephone channel bit rate	b x d = 8000/s x 8 bit = 64kbit/s

TS0-ÇTB Bulma Algoritması

Soru: Veri haberleşmesi, ikili sayı sistemi (bit: 0/1). Sürekli 1 ve 0ların transfer edildiği bir haberleşme ortamında alıcı taraf kanalları nasıl belirleycek?

Çerçeve Tantım Bilgisi (FAW), 125mikro saniyede bir 8 bit olarak gelir. Amaç seri olarak gelen 1 veya 0 değerlerinden doğru çerçeve tanıtım bilgisini belirlemektir.



Kanalların yerleri zaman domeninde belirlenmiş olur.
TS: Time Slot – Bir aboneye ait kanal
ÇTB: Çerçeve Tanıtım Bilgisi

Signalling Multiframe

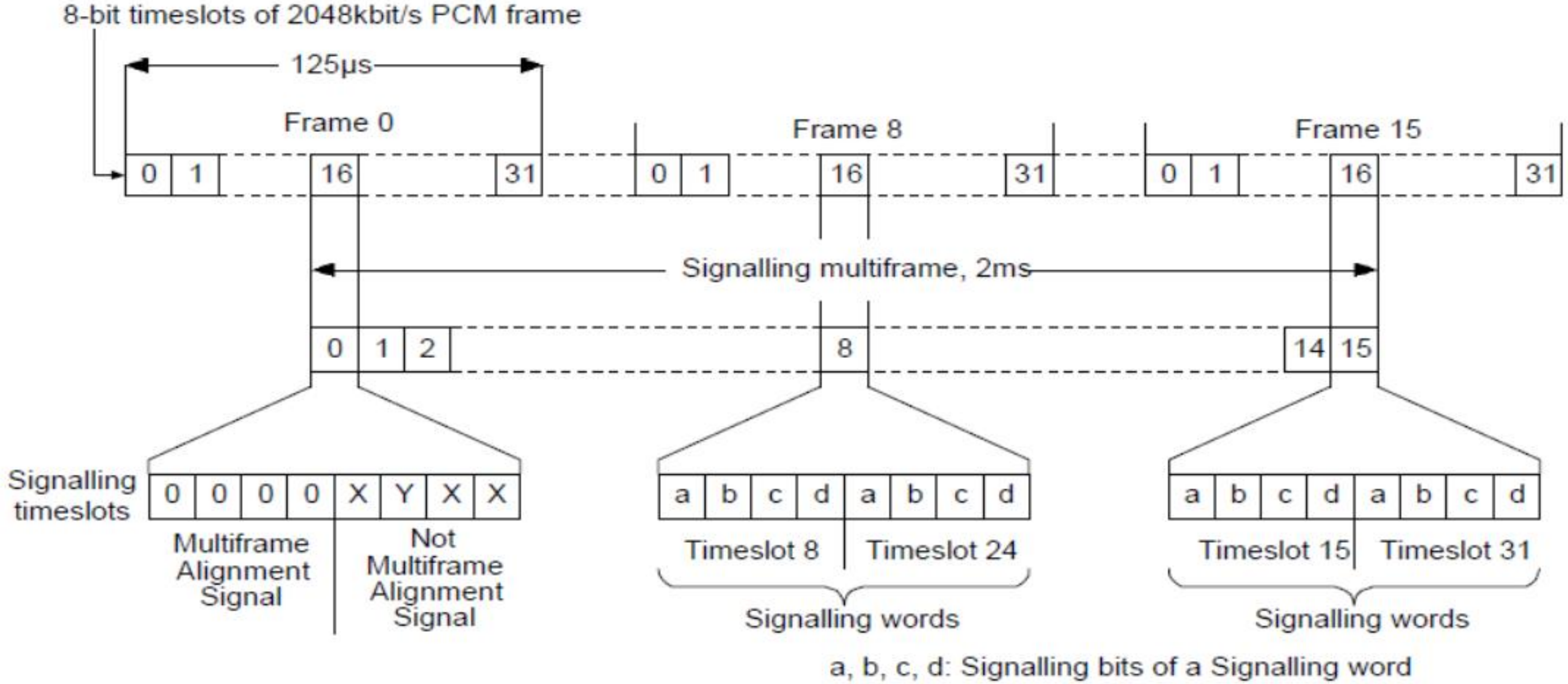


Figure 9: Signalling multiframe

The first four bits in timeslot 16 of the first frame (frame 0) of the signalling multiframe are used to transmit the Multiframe Alignment Signal (MFAS) = 0 0 0 0. The last four bits contain the Not Multiframe Alignment Signal (NMFAS) = X Y X X. The signalling multiframe structure is as shown in the table below:

frame number	Bits in channel timeslot 16	
	a b c d	a b c d
0	0 0 0 0	X Y X X
1	Telephone channel 1	Telephone channel 16
2	Telephone channel 2	Telephone channel 17
3	Telephone channel 3	Telephone channel 18
4	Telephone channel 4	Telephone channel 19
5	Telephone channel 5	Telephone channel 20
6	Telephone channel 6	Telephone channel 21
7	Telephone channel 7	Telephone channel 22
8	Telephone channel 8	Telephone channel 23
9	Telephone channel 9	Telephone channel 24
10	Telephone channel 10	Telephone channel 25
11	Telephone channel 11	Telephone channel 26
12	Telephone channel 12	Telephone channel 27
13	Telephone channel 13	Telephone channel 28
14	Telephone channel 14	Telephone channel 29
15	Telephone channel 15	Telephone channel 30

0 0 0 0 = multiframe alignment signal

X = reserved bit normally set to 1;   Y = distant multiframe alarm bit

Table 2: Assignment of bits in timeslot 16 of a signalling multiframe for channel- associated signalling

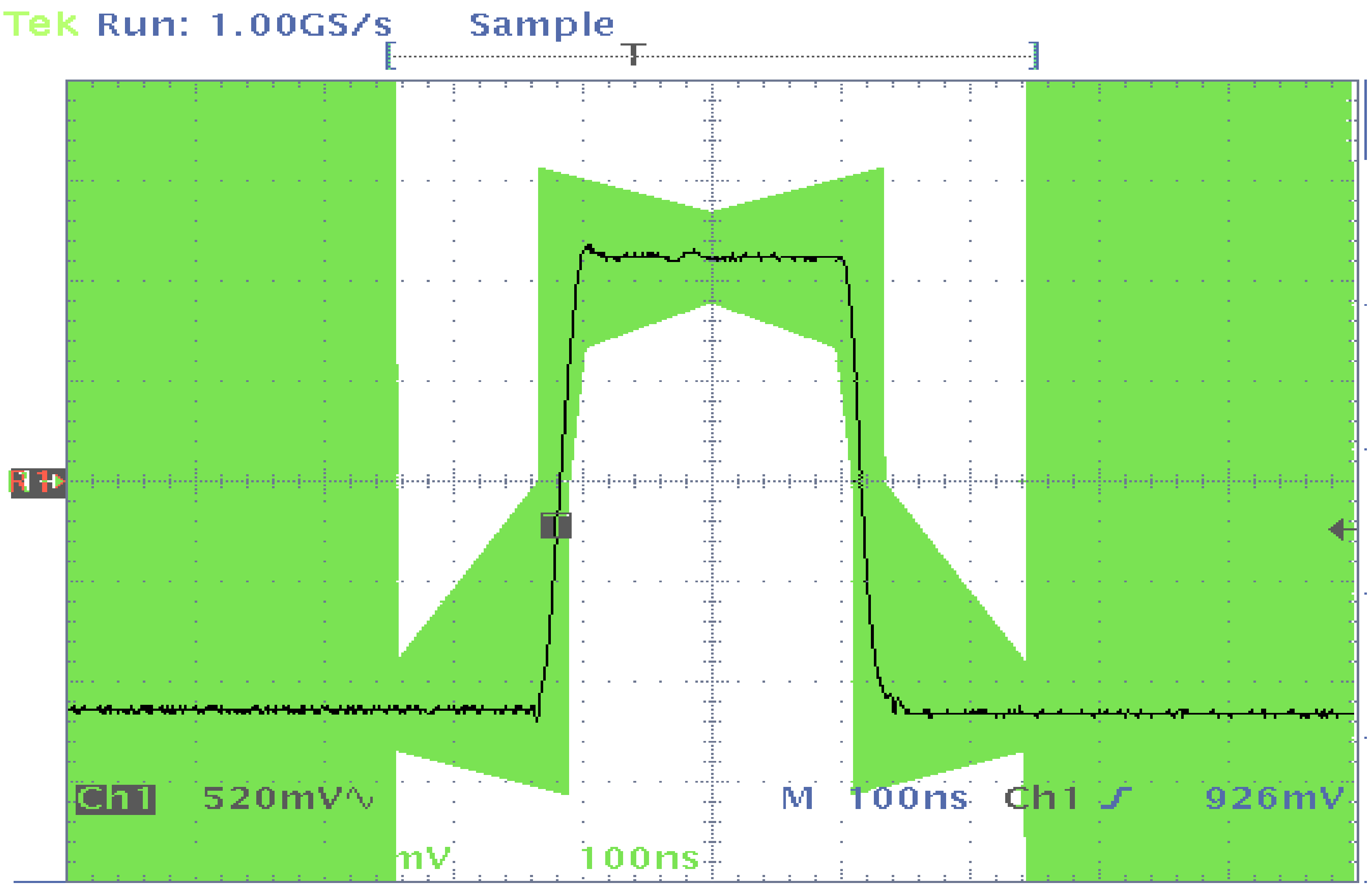
E1

- <u>**A Frame;**</u>
- **An E1 system consists of 32 time slots which contain the 64 kbit/s consecutive samples. This is called a frame.**
- <u>**The Transmission Rate;**</u>
- **When combined together 32 time slots x 64 Kbits/s = 2048 Kbits/s which is the speed of the E-1 transmission system.**
- <u>**ITU-T Specifications;**</u>
- **ITU-T Rec. G.703; Sayısal sinyaller haberleşme ortamından iletilmez. Neden? Zayıflar, gürültü. HDB3 formatında analog sinyal olarak iletilir.**
  - Bit rate: 2048 Kbit/s +/- 50 ppm
  - Code: HDB3 (high density bipolar 3)

Table 6/G.703

Pulse shape (normally rectangular)	All marks of a valid signal must conform with the mask (see figure 15/G.703) irrespective of the sign. The value V corresponds to the peak value.	
Pair(s) in each direction	One coaxial pair	One symmetrical pair
Test load impedance	75 ohms resistive	120 ohms resistive
Nominal peak voltage of a mask (pulse)	2.37 V	3 V
Peak voltage of a space (no pulse)	0 ± 0.237 V	0 ± 0.3 V
Nominal pulse width	244 ns	
Ratio of the amplitude of positive and negative pulses at the centre of the pulse interval	0.95 to 1.05	
Ratio of the widths of positive and negative pulses at the normal half amplitude	0.95 to 1.05	
Maximum peak-to peak jitter at an output port	Refer to § 2 of Recommendation G.823	

E1

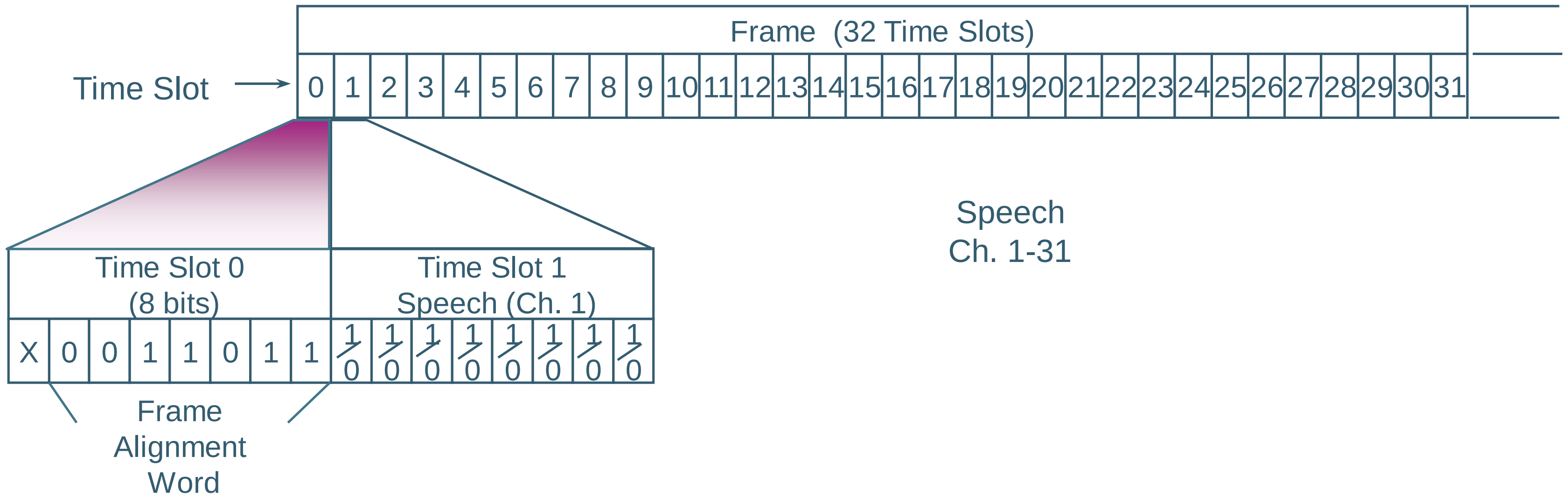


ITU-T Specifications;

- **ITU-T Rec. fig.15/G.703**

  sample capture of an E1 signal with the pulse shape mask

ITU-T Rec. G.704



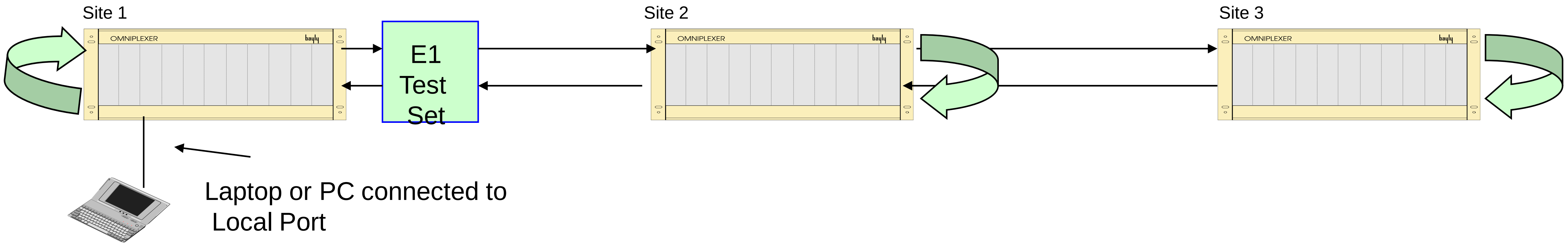
Applying this framing method to the OMNIBranch and OMNIFlex.

- **TS 0 is used for framing and alarm information**
- **The OMNIBranch assigns channel numbers 1 to 31, for usable transmission.**

CCITT G.704 (32 Time Slots)																															
0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31

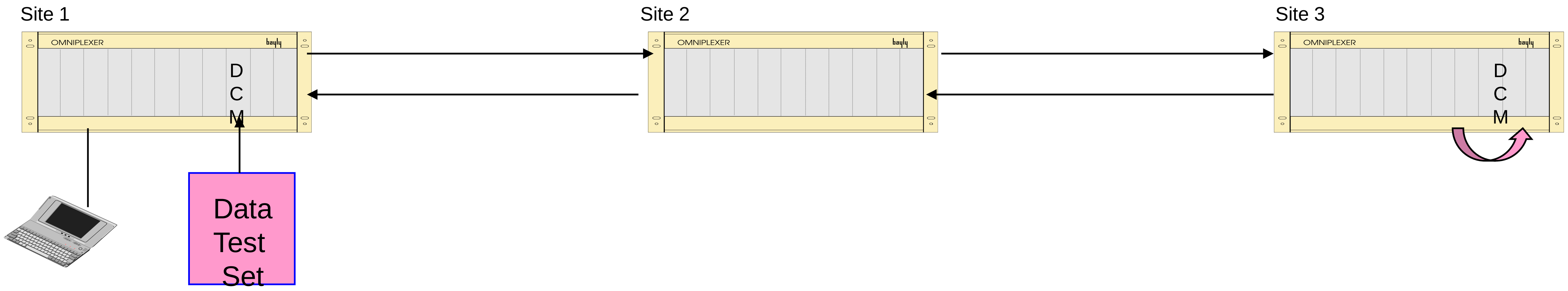
OMNIBranch / OMNIFlex - 31 Time Slot Assignments																															
-	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31

E1 Bit Error Rate Testing



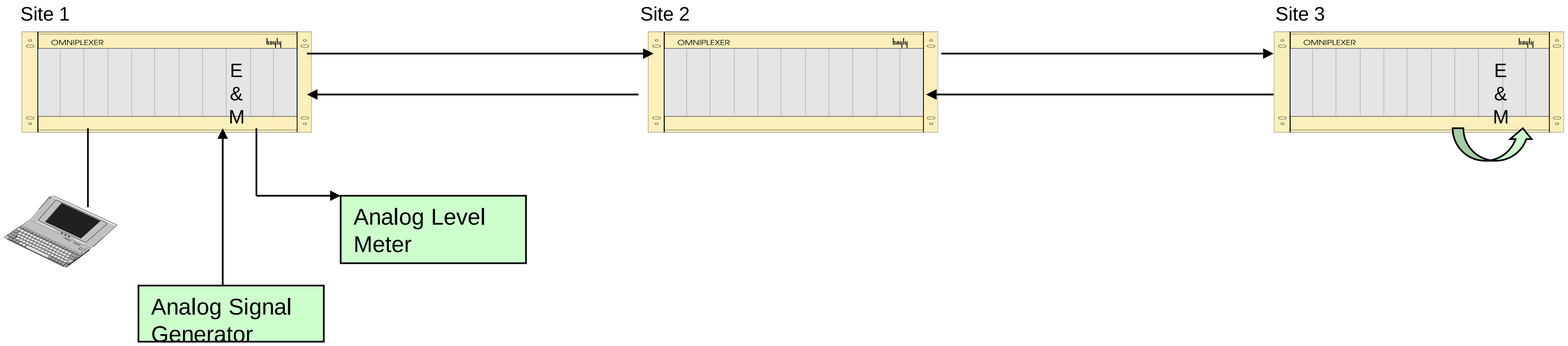
- From one location, technician can selectively test each link
- E1 remote loopback enabled sequentially to test link-by-link
- Testing performed in both “West” and “East” directions (using drop/insert E1 test set)
- Can test entire E1 bandwidth or time slot increments
- Duration of test is ½ to 1 hour
- Error rate of 10E-6 is acceptable, error rate of 10E-8 to 10E-9 is expected
- OMNIPlexer’s built-in E1 error rate testing will raise alarm if error rate rises above pre-programmed threshold (10E-6 to 10E-9)

Data Channel Testing



- From one location, technician can perform end-to-end bit error rate test for each data circuit
- Data channel remote loopback enabled
- Can test entire one time slot, multiple time slots or sub-rate time slot data channels
- Duration of test is ½ to 1 hour
- Error rate of 10E-6 is acceptable, error rate of 10E-8 to 10E-9 is expected

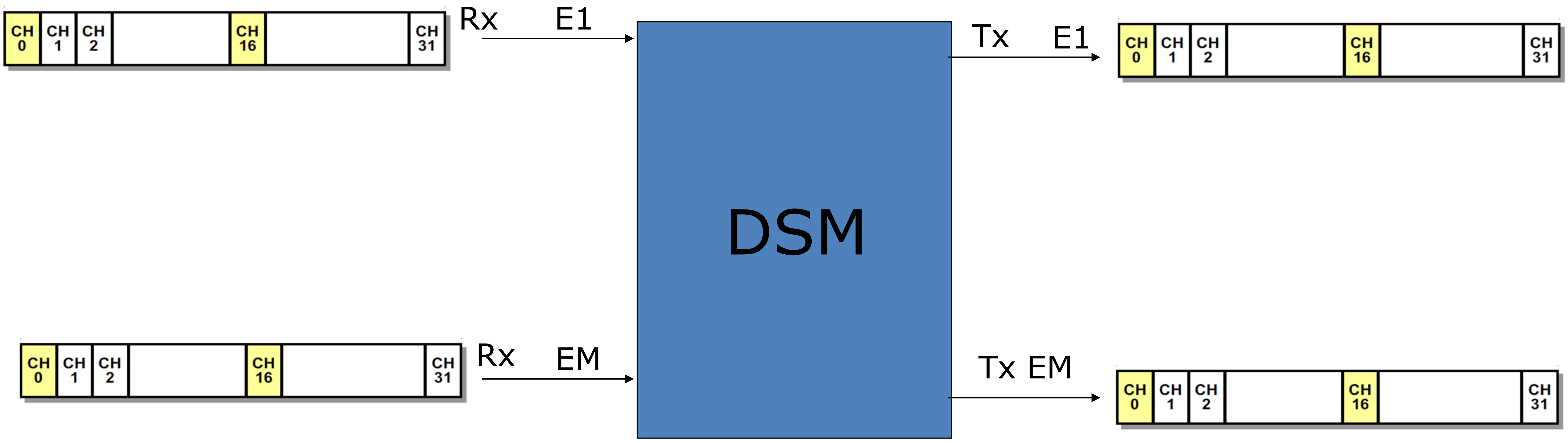
Voice Channel Testing



- From one location, technician can perform end-to-end bit test for each analog circuit
- Analog channel remote loopback enabled
- Measure analog level, signal-to-noise ratio, noise floor, distortion

Digital Switching Matrix

Digital Switching Matrix (DSM) ve E1 devresinde kanal anahtarlama



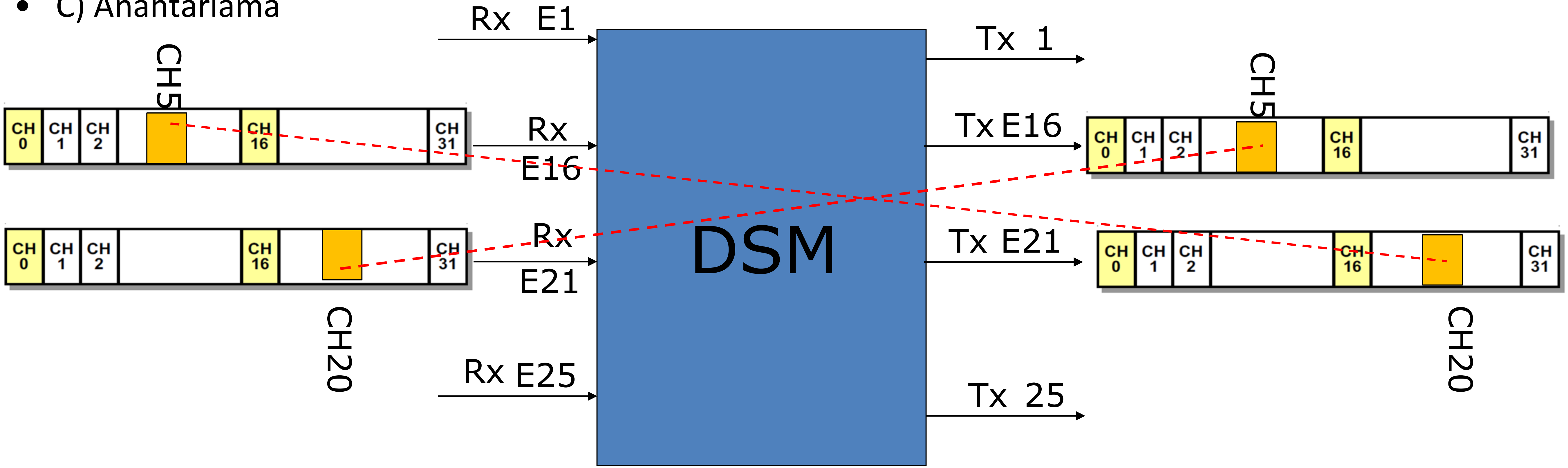
Örnek: Toplam E1 sayısı 25 olan DSM de 455 nolu aboneyi 620 nolu aboneye anahtarlayın

- A) Aynı anda hizmet alan toplam abone sayısı=25 x 30 =750
- B) 455 nolu ve 620 nolu aboneler kanalı hangi E1 de?

  455/30=15. …; 16 nolu E1'in 5 nolu kanalı (Time Slot); Time Slot=455-(16-1)*30=5

  620/30=20. …; 21 nolu E1'in 20 nolu kanalı (Time Slot); Time Slot=620-(21-1)*30=20

- C) Anahtarlama

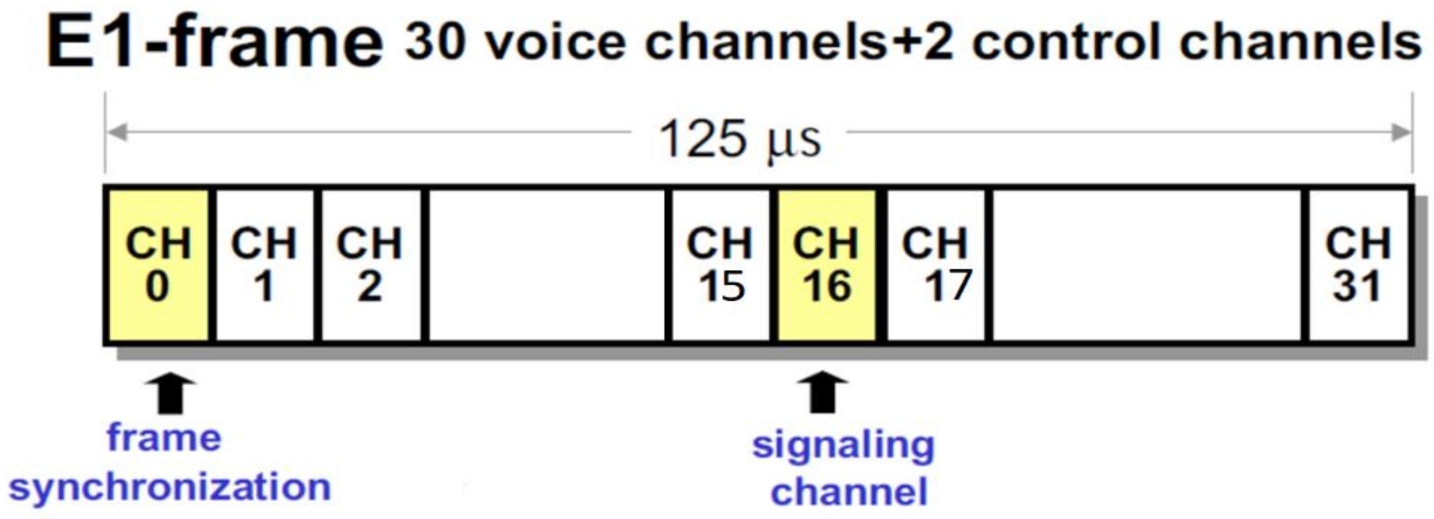


Örnek: Toplam E1 sayısı 5 olan DSM de 75 nolu aboneyi 90 nolu aboneye anahtarlayın

- A) Toplam aynı anda hizmet alacak abone sayısı=5 x 30 =150
- B) E1 devrelerine ilişkin Tx, Rx Kanal Sıralaması yapınız
  - 1. E1 Kanal Sıralama: 1…30
  - 2. E1 Kanal Sıralama: 31…60
  - 3. E1 Kanal Sıralama : 61… 90
  - 4. E1 Kanal Sıralama : 91 … 120
  - 5. E1 Kanal Sıralama : 121 … 150

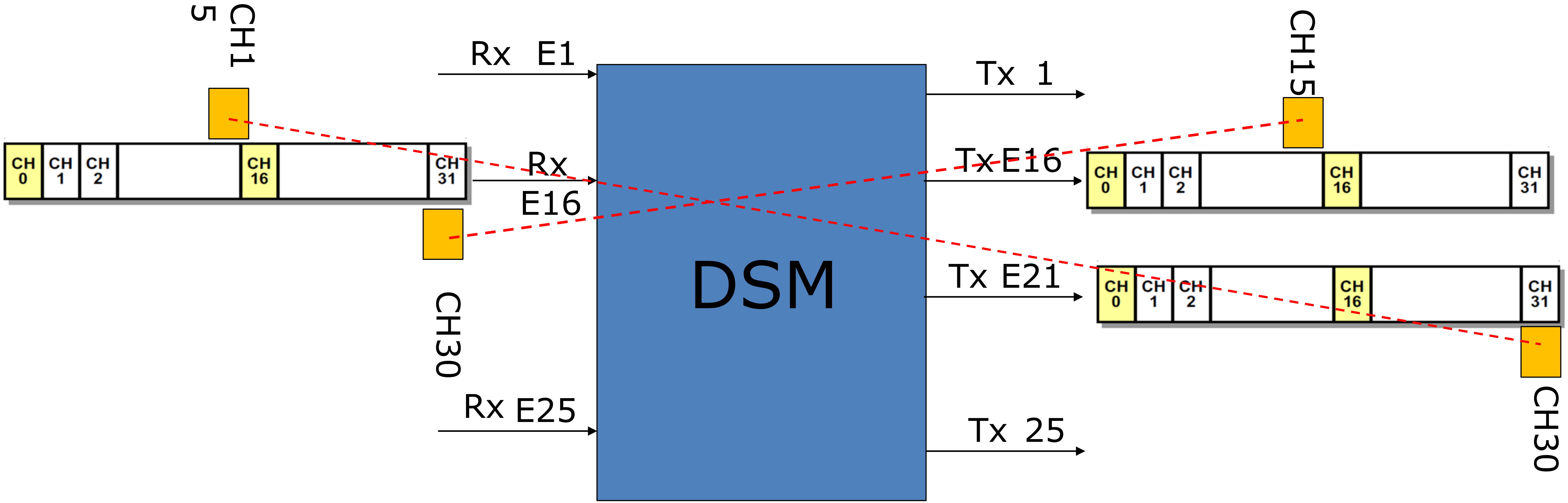
C) Anahtarlanacak abonelerin konumlarını belirleyin

- E1 data kanalları (30 adet)
- E1 data kanal sıralama: 1..15, 17..31 (0. Kanal: FAW, 16. Kanal: Çoklu Çerçeve tanıtım bilgisi + Sinyalleşme bilgileri)
- 75 nolu abone = 30*2 +15: 3. E1'in 15 nolu kanalı
- 90 nolu abone=3*30: 3.E1'in 31 nolu kanalı



Örnek: Toplam E1 sayısı 5 olan DSM de 75 nolu aboneyi 90 nolu aboneye anahtarlayın

- D) DSM anahtarlama devresini çiziniz.

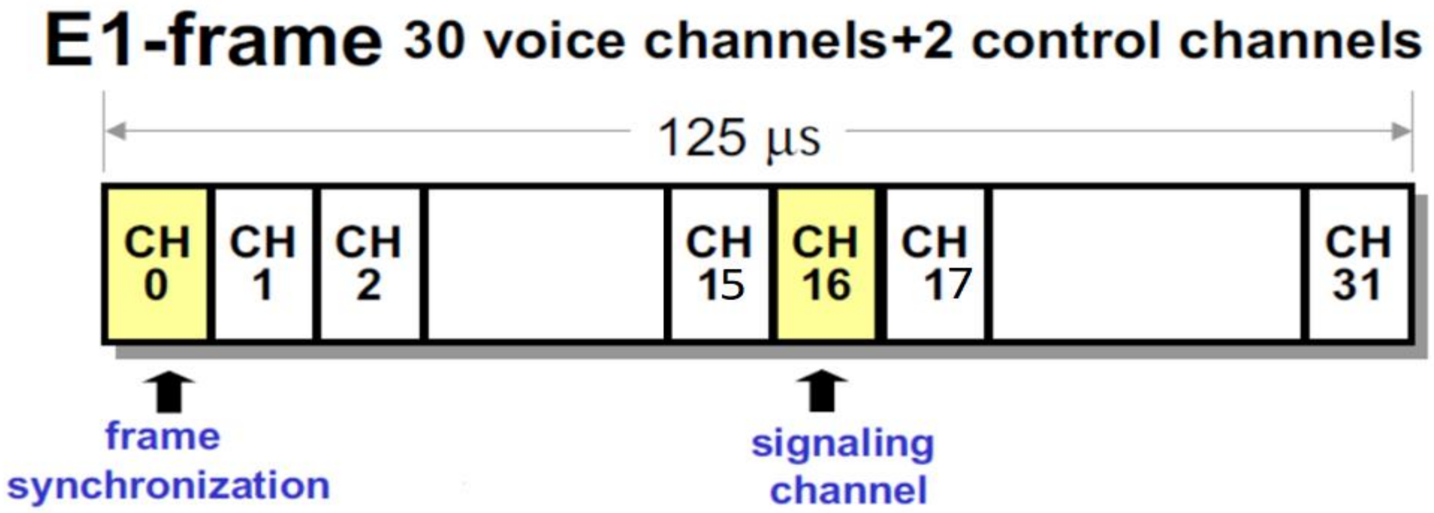


Örnek: Toplam E1 sayısı 8 olan DSM de 6. E1 10. Kanalı 4.E1 in 29. Kanalına anahtarlanmıştır. DSM anahtarlama devresini çiziniz. Abone numaralarını bulunuz

- A) Abone numaralarını bulunuz.
- Aynı anda hizmet alacak toplam abone sayısı: 8 x 30=240
- Kanal Sıralaması:

  1. E1 Kanal Sıralama: 1…30
  - 2. E1 Kanal Sıralama: 31…60
  - 3. E1 Kanal Sıralama : 61… 90
  - 4. E1 Kanal Sıralama : 91 … 120
  - 5. E1 Kanal Sıralama : 121 … 150
  - 6. E1 Kanal Sıralaması: 151 … 180
  - 7. E1 Kanal Sıralaması: 181 … 210
  - 8. E1 Kanal Sıralaması: 211 … 240

- 6. E1 10.Kanal: 5 x 30 + 10=150 +10=160
- 4. E1 29. Kanal: 3 x 30 +30 (29+1) =120 (ÇÇTB: 16. Kanal)



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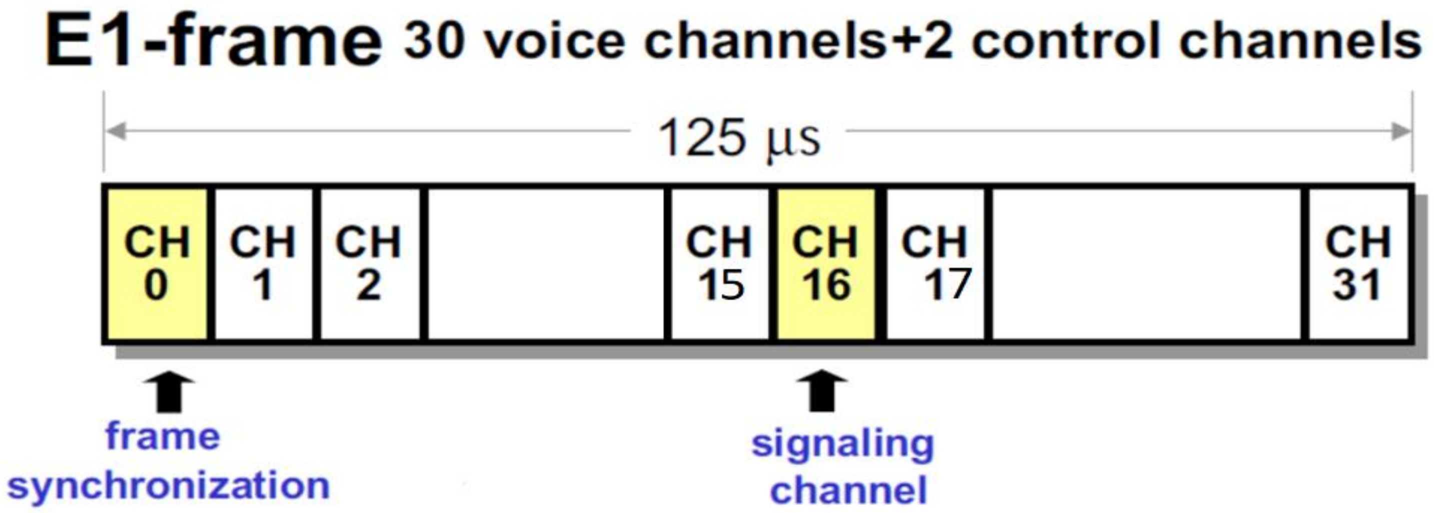

Arayan ve Aranan Abonelerin E1 ve Kanal Numaralarını Bulan Algoritma

Adım-1: Aramaların hangi E1 devresi üzerinden olduğunu bulmak için numara 30 abone sayısına bölünür.
- Arayan abone, A1=tamsayı(arayan abone numarası/30),
- Aranan abone, A2=tamsayı(aranan abone numarası/30),

Adım-2: K1=Arayan Abone Numarası-30*A1, K2=Arayan Abone Numarası-30*A2.

Adım-3:
- Eğer K1=0 ise M1=A1, Kanal Numarası, B1=31 alınır.
- Eğer K2=0 ise M2=A2, Kanal Numarası, B2=31 alınır.
- Git "Adım-5"

Adım-4:
- Konuşma kanallarının E1 üzerindeki dağılım 1.kanal ile 15. Kanal ve 17.Kanal ile 31.Kanal arasındadır. Kanal Çerçeve Tanıtım bilgisi, 16.Kanaldan ise Çoklu Çerçeve Tanıtım ve İşaretleşme Bilgileri gönderilir.
- Eğer K1≠ 0 ise M1=A1+1 alınır.
- 1<=K1<=15 arasında ise Abone kanal numarası, B1=K1 alınır.
- 16<=K1<=29 arasında ise Abone kanal numarası, B1=K1+1 alınır.
- Eğer K2≠ 0 ise M2=A2+1 alınır.
- 1<=K2<=15 arasında ise Abone kanal numarası, B2=K2 alınır.
- 16<=K2<=29 arasında ise Abone kanal numarası, B2=K2+1 alınır.

Adım-5:
- Anahtarlama işlemi tamamlanmıştır. Arayan abone M1 inci E1 devresini B1 inci kanalı üzerinden aradığı abonesi ile M2 inci E1 devresinin B2 inci kanalı üzerinden anahtarlama yapılarak iletişim kurmaktadır.

Örnek:

Telefon numarası 4718 olan abone telefon numarası 2057 olan aboneyi arıyor. Anahtarlama DSM (Digital Switching Matrix) sistemindeki E1 devreleri üzerinden yapılmaktadır. Bir E1 devresinde 30 konuşma kanalı bulunmaktadır.

a) Adım-1'deki ifadelerden arayan ve aranan aboneler için A1,A2 değerlerini bulunuz.

- Arayan abone, A1=tamsayı(arayan abone numarası/30) = tam sayı(4718/30) = 157
- Aranan abone, A2=tamsayı(aranan abone numarası/30) = tam sayı(2057/30) = 68

b) Adım-2'deki ifadelerden K1 ve K2 değerlerini hesaplayın.

- K1=Arayan Abone Numarası-30*A1 = 4718 – 30*157 =4718-4710 =8
- K2=Arayan Abone Numarası-30*A2 =2057 – 30*68=2057-2040 =17

c) Eğer K1=0 veya K2=0 ise Adım-3 çalıştırın.

d) Eğer K1≠ 0 veya K2≠ 0 ise Adım-4 çalıştırın.

- Eğer K1≠ 0 ise M1=A1+1=158 alınır.
- 1<=K1<=15 arasında ise Abone kanal numarası B1=K1 alınır. B1=8

- Eğer K2≠ 0 ise M2=A2+1=69 alınır.
- 16<=K2<=29 arasında ise Abone kanal numarası, B2=K2+1=18 alınır.

e) Adım-5'i çalıştırın. Bulduğunuz M1,M2, B1, B2 değerlerini yerine koyarak yorum yapınız.

- Anahtarlama işlemi tamamlanmıştır. Arayan abone 158 inci E1 devresini 8 inci kanalı üzerinden aradığı abonesi ili 69 inci E1 devresinin 18 inci kanalı üzerinden anahtarlama yapılarak iletişim kurmaktadır.

Örnek:

Telefon numarası 2728 olan abone telefon numarası 1055 olan aboneyi arıyor. Anahtarlama DSM (Digital Switching Matrix) sistemindeki E1 devreleri üzerinden yapılmaktadır. Bir E1 devresinde 30 konuşma kanalı bulunmaktadır.

a) Adım-1'deki ifadelerden arayan ve aranan aboneler için A1,A2 değerlerini bulunuz.

- Arayan abone, A1=tamsayı(arayan abone numarası/30) = tam sayı(2728/30) = 90
- Aranan abone, A2=tamsayı(aranan abone numarası/30) = tam sayı(1055/30) = 35

b) Adım-2'deki ifadelerden K1 ve K2 değerlerini hesaplayın.

- K1=Arayan Abone Numarası-30*A1 = 2728 – 30*90 =2728-2700 =28
- K2=Arayan Abone Numarası-30*A2 =1055 – 30*35=1055-1050 =5

c) Eğer K1=0 veya K2=0 ise Adım-3 çalıştırın.

d) Eğer K1≠ 0 veya K2≠ 0 ise Adım-4 çalıştırın.

- Eğer K1≠ 0 ise M1=A1+1=91 alınır.
- 1<=K1<=15 arasında ise Abone kanal numarası B1=K1 alınır. B1=29

- Eğer K2≠ 0 ise M2=A2+1=36 alınır.
- 16<=K2<=29 arasında ise Abone kanal numarası, B2=K2+1=6 alınır.

e) Adım-5'i çalıştırın. Bulduğunuz M1,M2, B1, B2 değerlerini yerine koyarak yorum yapınız.

- Anahtarlama işlemi tamamlanmıştır. Arayan abone 91 inci E1 devresini 29 inci kanalı üzerinden aradığı abonesi ili 36 inci E1 devresinin 6 inci kanalı üzerinden anahtarlama yapılarak iletişim kurmaktadır.

Örnek

Telefon numarası 4718 olan abone telefon numarası 2057 olan aboneyi arıyor. Anahtarlama DSM (Digital Switching Matrix) sistemindeki E1 devreleri üzerinden yapılmaktadır. Bir E1 devresinde 30 konuşma kanalı bulunmaktadır.

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- Arayan abone, A1=tamsayı(arayan abone numarası/30) = tam sayı(4718/30) = 157
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- K1=Arayan Abone Numarası-30*A1 = 4718 – 30*157 =4718-4710 =8
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c) Eğer K1=0 veya K2=0 ise Adım-3 çalıştırın.

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- Eğer K1≠ 0 ise M1=A1+1=158 alınır.
- 1<=K1<=15 arasında ise Abone kanal numarası B1=K1 alınır. B1=8

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- http://www.ee.cityu.edu.hk/~csl/sigana/sig01.ppt
- İnternet ortamından sunum ve ders notları

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Sincerely,

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thank you