NGN

next generation network

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About the Tutorial

Next Generation Networks (NGN) is a part of present-day telecommunication system, which is equipped with capabilities to transport all sorts of media, such as voice, video, streaming audio/video, text, etc. NGN is developed around the concept of packet switching as in Internet Protocol architecture. It is more efficient and equally complicated and involves number of systems, equipment, and processing.

Audience

This tutorial is developed for beginners to help them understand the basics of NGN and its components. . After finishing this tutorial, you would acquire a good know-how of Next Generation Networks.

Prerequisites

This tutorial requires a basic understanding of computer networking, signal processing, and telecommunication system. To get the most of this tutorial, the readers are highly encouraged to learn the required concepts first.

Disclaimer & Copyright

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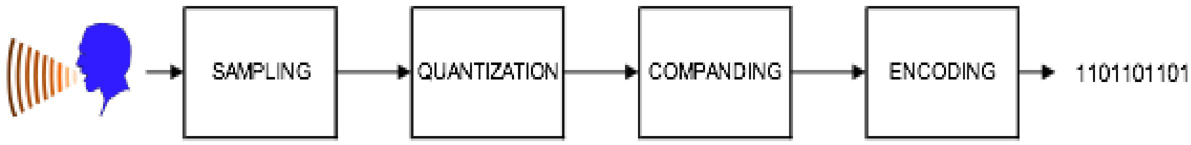
1. NGN – PULSE CODE MODULATION

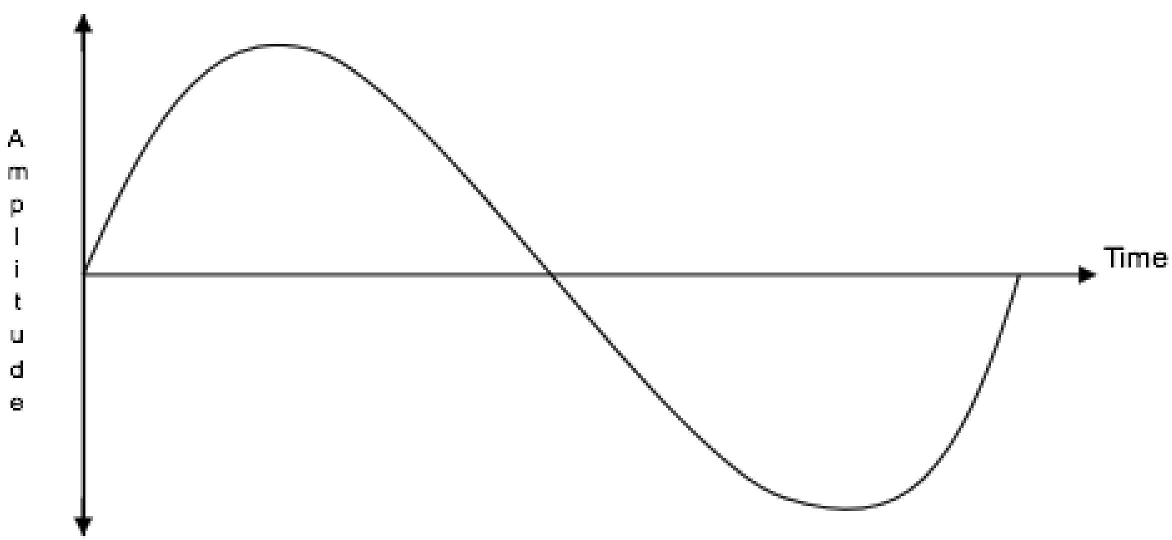
The advent of high-speed voice and data communications has brought about the need for a fast medium for transporting the information. Digital circuits or links have evolved from the need to transmit voice or data in digital form.

The conversion from analogue to digital form follows a four-stage processes (*see the following Figure*) and will be detailed in the following sections.



Sampling

Voice frequencies take the form of an analogue signal i.e. sine wave (*see the following Figure*). This signal has to be converted into a binary form for it to be carried over a digital medium. The first stage of this conversion is to convert the audio signal into a **Pulse Amplitude Modulation**(PAM) signal. This process is generically known as **sampling**.

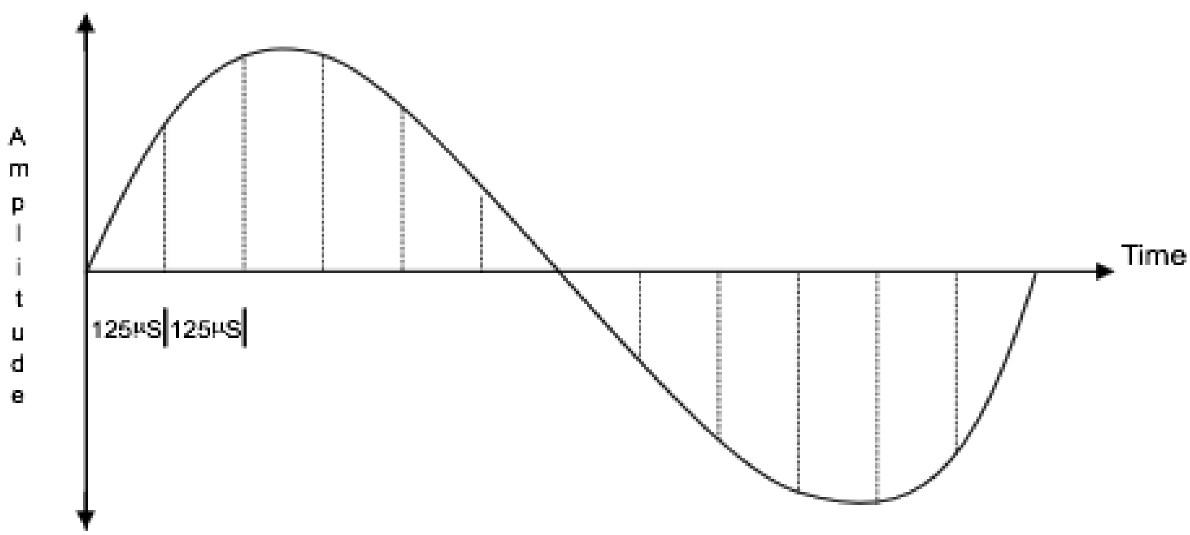


The sampling process must gather sufficient information from the incoming voice frequencies to enable a copy of the original signal to be made. Voice frequencies are normally in the range of **300Hz to 3400Hz**, typically known as the **commercial speech band**.

To obtain a sample, a sampling frequency is applied to the original voice frequency. The sampling frequency is determined by the **Nyquist Sampling Theorem**, which dictates that **"the frequency of sampling should be at least twice the highest frequency component**."

This ensures that a sample is taken a minimum of once in each half cycle, thus, eliminating the possibility of sampling at zero points of the cycle, which would have no amplitude. This results into the sampling frequency being a minimum of 6.8 KHz.

The European standard samples an incoming signal at **8 KHZ**, ensuring a sample, is taken every **125micro seconds** or 1/8000th of a second (*see the following Figure*).

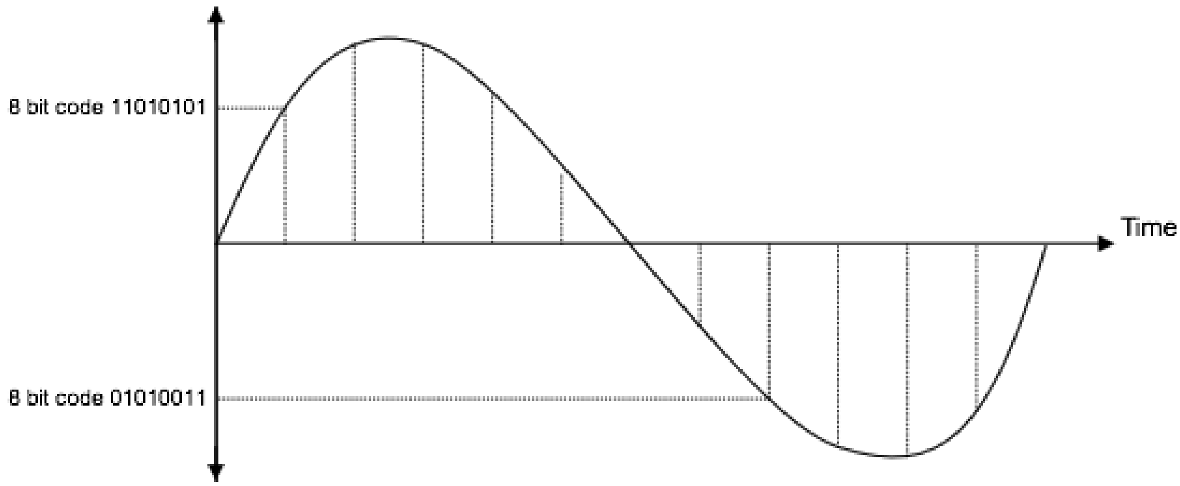


Quantization

The amplitude of each sample would ideally be assigned a binary code (1's or 0's), but as there can be an infinite number of amplitudes; therefore, there need to be an infinite number of binary codes available. This would be impractical, so another process has to be employed, which is known as **quantizing.**

Quantizing compares the PAM signal against a quantizing scale, which has a finite number of discrete levels. The quantizing scale splits into 256 quantizing levels, of which, 128 are positive levels and 128 are negative levels.

The quantization stage involves allocating a unique 8 bit binary code appropriate to the quantizing interval into which the amplitude of the PAM signal falls (*see the following Figure*).



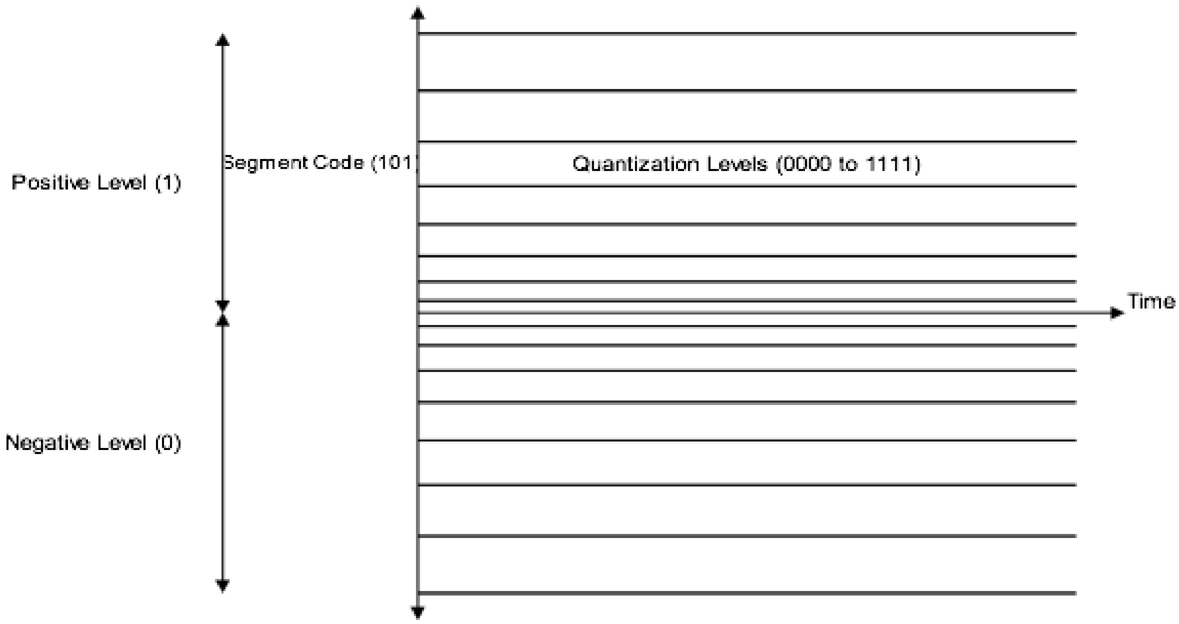
This comprises of 1 polarity bit with the remaining 7 bits used to identify the quantization level (*as shown in the above figure*).

The first bit as seen before is the polarity bit, the next three bits for the segment code, giving eight segment codes, and the remaining four bits for the quantization level, giving sixteen quantization levels.

Companding

The quantizing process itself leads to a phenomenon known as **quantization distortion**. This occurs when the sampled signal amplitude falls between the quantization levels. The signal is always rounded up to the nearest whole level. This difference between the sampled level and the quantizing level is quantizing distortion.

The rate of change of the amplitude of a signal varies at different parts of the cycle. This happens most at high frequencies as the amplitude of the signal changes faster than at the low frequencies. To overcome this, the first segment code has the quantization levels close together. The next segment code is then double the height of the previous and so on. This process is known as **companding**, as it compresses larger signals and expands smaller signals.



In Europe they use the **A-law** of companding, compared to North America and Japan who use the **μ law**.

As quantization distortion is equivalent to noise, companding improves the signal to noise ratio on low amplitude signals, and produces an acceptable signal to noise ratio over the complete range of amplitudes.

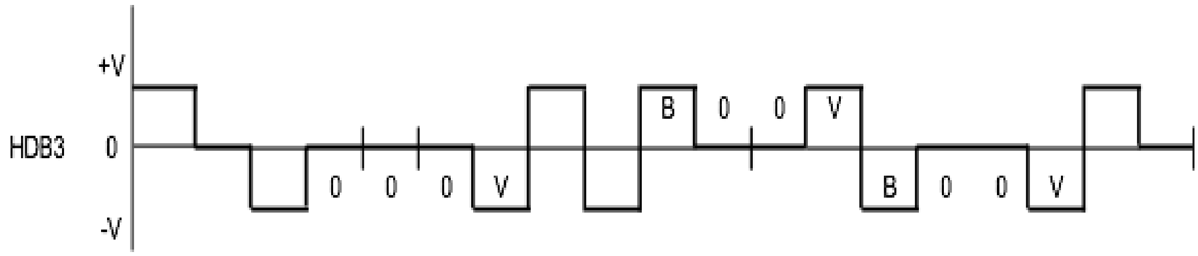
Encoding

In order for the binary information to be transmitted over a digital path, the information has to be modified into a suitable line code. The encoding technique employed in Europe is known as **High Density Bipolar 3 (HDB3).**

HDB3 is derived from a line code called AMI or **Alternate Mark Inversion**. Within AMI encoding, there are 3 values used: no signal to represent a binary 0, and a positive or negative signal that is used alternately to represent a binary 1.

One problem associated with AMI encoding occurs when a long string of zeros are transmitted. This can cause phase lock loop problems at the distant end receiver.

HDB3 works in a similar way to AMI, but incorporates an extra encoding step that replaces any string of four zeros by three zeros followed by a 'violation bit.' This violation is of the same polarity of the previous transition (*see the following Figure*).



As can be seen in the example, 000V replaces the first string of four zeros. However, using this type of encoding could lead to a mean D.C. level being introduced into the signal, as a long string of zeros could be present, all being encoded in the same way. To avoid this, the encoding of each successive four zeros is changed to B00V, by using a 'Bipolar violation' bit that alternates in polarity.

From this, it can be assumed that with HDB3 encoding, the maximum number of zeros without a transition is three. This encoding technique is often referred as the **modulation format.**

2. NGN – MULTIPLEXING

Multiplexing

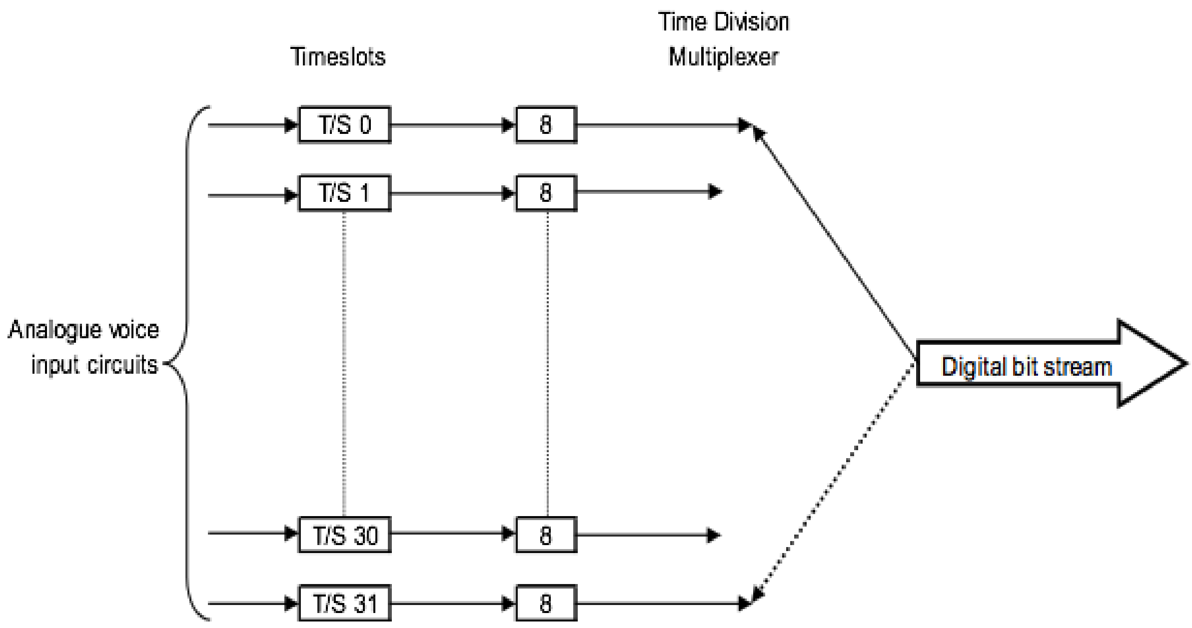
So far, we have been concentrating only on one voice channel. Now, we need to combine a number of these channels into a single transmission path, a process known as **multiplexing**. Multiplexing is a process employed whereupon several channels can be combined, in order for them to be transmitted over a single transmission path. The process commonly in use in telephony is known as **Time Division Multiplexing (TDM)**.

As we have seen before, sampling for one channel takes place every **125 micro seconds**. This makes it possible to sample other channels during this period. In Europe, the time span is divided into **32** time periods, known as **timeslots**. These 32 timeslots can then be grouped together to form a **frame** (*see the following figure*).

Consequently, the time duration of a frame can be considered as 125micro seconds. It can now also be assumed that as each timeslot consists of 8 data bits, and is repeated 8000 times a channel rate of 64000 bits per second or 64Kbits is attainable. With this information it is now possible to determine the total number of data bits transmitted over the single path, known as the **system bit rate**. This is calculated using the following formula:

System bit rate = Sampling frequency x Number of timeslots x Bits per timeslot

= 8000 x 32 x 8 = 2048000 bits/sec = 2.048 Mbits



Of the 32 channels available, 30 are used for speech transmission, and the remaining 2 timeslots are used for alignment and signalling. The following section will explain the function of all the timeslots.

3. NGN – FRAME STRUCTURE

Timeslot 1 to 15 and 17 to 31

These 30 timeslots are available for the transmission of the digitized analogue signal in 8-bit form, with a bandwidth of 64 kbit/s (e.g. the customers' data).

Timeslot 0

The European recommended system defines that the Timeslot 0 of each frame is used for synchronization, also known as **frame alignment** (*see the following Figure*). This ensures that the timeslots in each frame are aligned between the transmitting station and the receiving station.

Frame Alignment Word T/S 0 Even Frames

Bit 1	Bit 2	Bit 3	Bit 4	Bit 5	Bit 6	Bit 7	Bit 8
C	0	0	1	1	0	1	1

C = Cyclic Redundancy Check (CRC)

The **frame alignment word** (FAW) is carried in data bits 2 to 8 of each even frame, while the odd frames carry a **not frame alignment word** (NFAW) in data bit 2 (*see the following Figure*).

Not Frame Alignment Word T/S 0 Odd Frames

Bit 1	Bit 2	Bit 3	Bit 4	Bit 5	Bit 6	Bit 7	Bit 8
C	1	X	*	*	*	*	*

C = Cyclic redundancy Check (CRC)
X = Far End Alarm
* = Spare

An error check is also available in timeslot 0, using a **cyclic redundancy check** (CRC) to verify the frame alignment, which is carried in data bit 1 of all frames. There is also the facility of reporting **Far End Alarms**, which is indicated by a binary 1 being inserted in data bit 3 of all the odd frames. The remaining data bits 4 to 8 of the odd frames can be utilized for national alarms and network management.

Timeslot 16

Timeslot 16 has 8 data bits available, and by using a variable code of 4 data bits, signalling can be performed for 2 voice channels in each frame.

Therefore it can be seen that 15 frames are required to complete the signalling for all the voice channels (*see the following Figure*).

Signalling Information T/S 16 Frames 1 to 15

Bit 1	Bit 2	Bit 3	Bit 4	Bit 5	Bit 6	Bit 7	Bit 8
a	b	c	d	a'	b'	c'	d'
Channel N				Channel N+15			

As there are now multiple frames being carried in a logical order, there has to be a device for aligning these. This is achieved by using the frame prior to the frames containing signalling information, known as Frame 0.

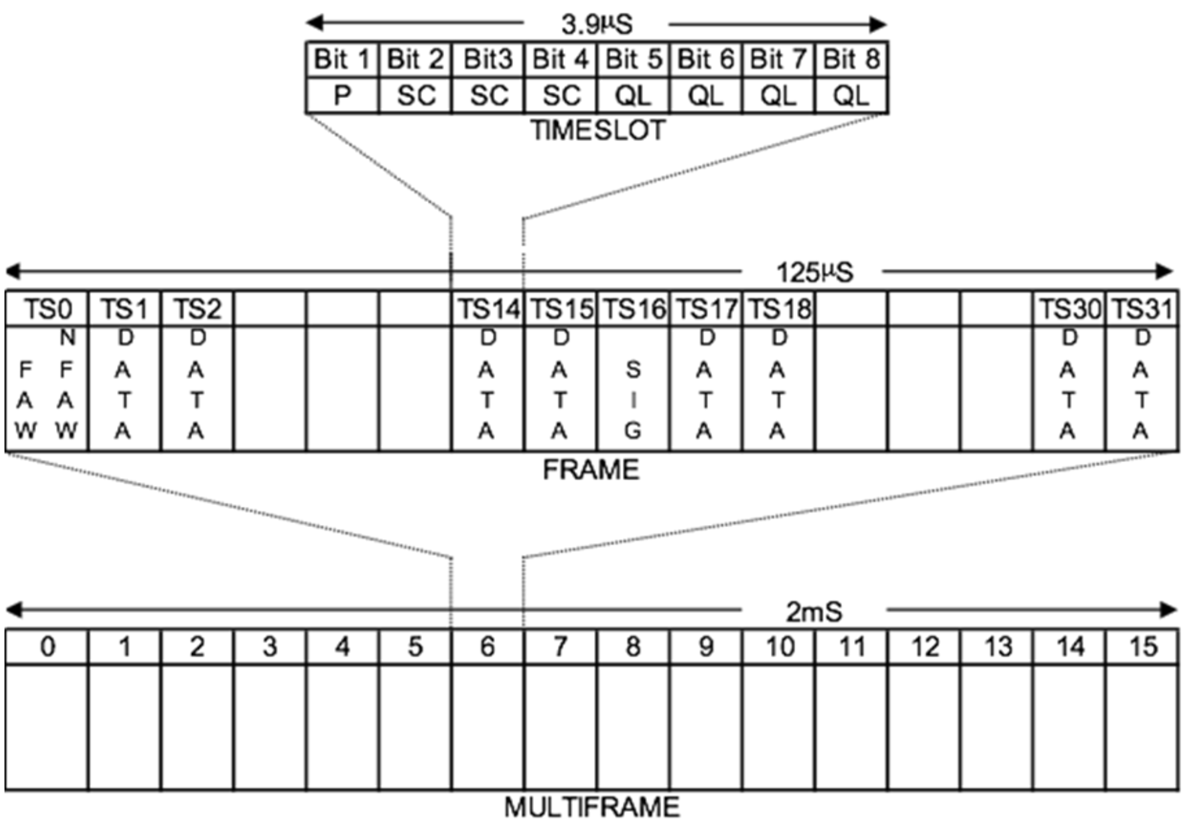
Timeslot 16 in Frame 0 contains a **multi-frame alignment word** (MFAW), using data bits 1 to 4, and are used to indicate the start of a multi-frame, which are checked at the receiving station (*see the following Figure*).

Multiframe Alignment Word T/S 16 Frame 0

Bit 1	Bit 2	Bit 3	Bit 4	Bit 5	Bit 6	Bit 7	Bit 8
0	0	0	0	*	Y	*	*

Y = Distant Multiframe Alignment Loss (DLMFA)
* = Spare

Data bit 6 can be used to indicate **distant multi-frame alignment loss** (DLMFA). As can be seen, a multi-frame consists of all the frames required to complete all speech and signalling operations, i.e. 16 frames, and is known as a **multi-frame** (*see the following Figure*).



The duration of a multi-frame can be calculated using the following:

Duration of multi-frame = Number of frames × duration of frame

$= 16 \times 125$ micro seconds

$= 2000$ micro seconds

$= 2$ milliseconds

The remaining channels are all usable for voice or data transmission, and are known as timeslots 1 to 15 and 17 to 31, and equate to channels numbered 1 to 30.

FAW = Frame Alignment Word

MFAW = Multi-frame Alignment Word

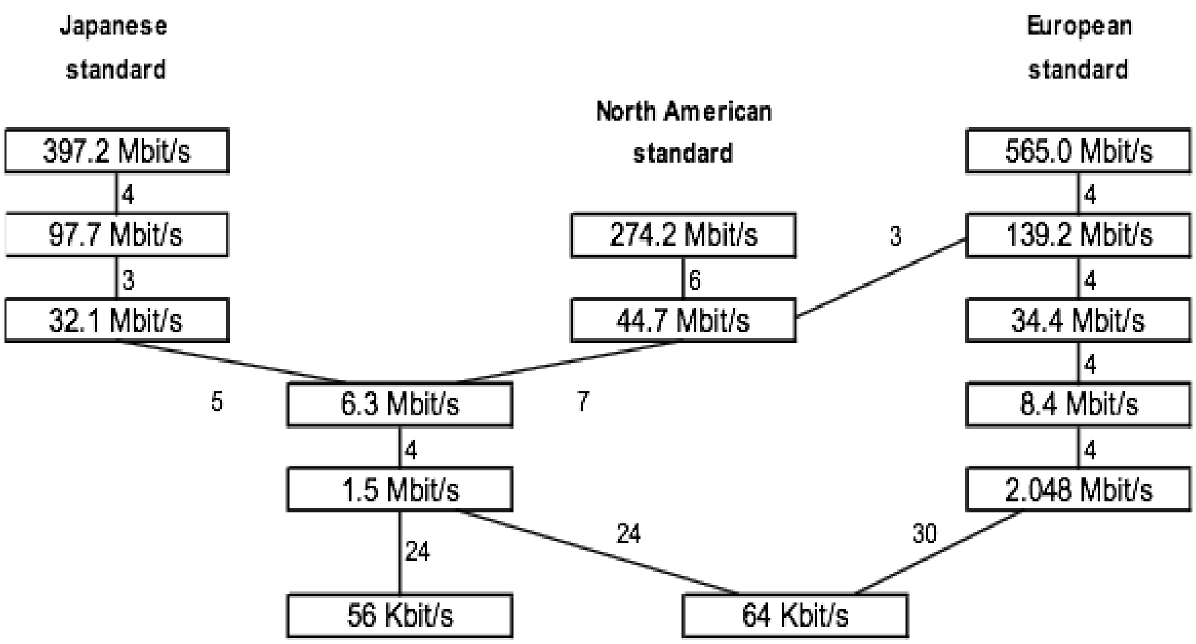
DATA = 8 bit data words

SIG = CAS signalling timeslot

4. NGN - HIGHER ORDER MULTIPLEXING

The Plesiochronous Digital Hierarchy (PDH) has been developed in stages from the basic 30-channel PCM (PCM-30) system.

As can be seen in the following *Figure*, there are three different hierarchical systems available, each supporting different line rates and multiplexing rates. The higher aggregate rates can therefore be achieved by grouping together the lower rates through the use of multiplexers.



The higher bit rate links also require additional bits for framing and control. For example, an 8.4 Mbits signal comprises of 4 × 2.048 Mbits = 8.192 Mbits, with the remaining 256 Kbits being used for framing and control.

The European and North American hierarchy systems are often referred by the letter '**E**' for European and '**T**' for North American, with the hierarchy levels being numbered consecutively. These hierarchy levels can be compared in the following *Figure*:

	Hierarchy Level	Bit Rate (Mbits)	Voice Channels
North America	T1	1.544	24
	T2	6.312	96
	T3	44.736	672
	T4	274.176	4032
European	E1	2.048	30
	E2	8.448	120
	E3	34.368	480
	E4	139.264	1920
	Not Defined	565.148	7680

These bit rates are often abbreviated to 1.5 meg, 3 meg, 6 meg, 44 meg, 274 meg and 2 meg, 8 meg, 34 meg, 140 meg, and 565 meg respectively.

As the legacy of PDH is so prominent in the telecommunications industry, it became necessary to accommodate these line rates in any new technology to be introduced, therefore many of the PDH line rates are supported by the Synchronous Digital Hierarchy (SDH). The only exception to this is the omission of the 8.4 Mbits level, which no longer has any practical meaning and is not supported by SDH.

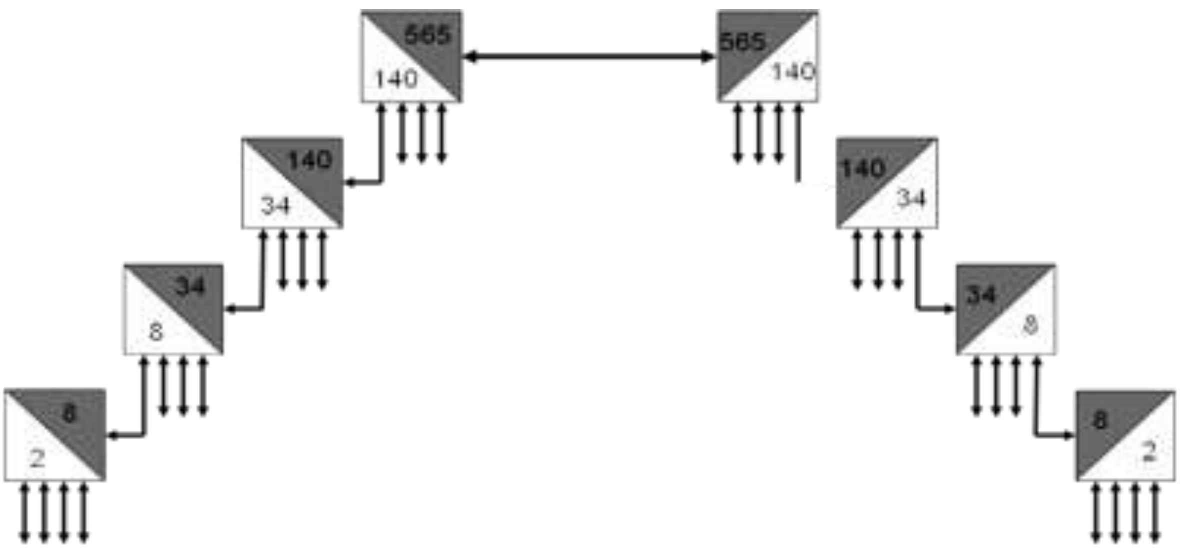
In the basic 2 Mbits system, the data is byte interleaved, whereby each 8-bit timeslot is sent one after the other. In the case of the higher hierarchy levels, the data streams are multiplexed together bit-by-bit. A disadvantage of this system is that the bit rate of each tributary signal can vary from the nominal value due to each multiplexer having their own independent clock supplies. These clock deviations are dependent on the line rate and can be compensated for by using justification techniques within the bandwidth remaining after the multiplexing stage. The line rate also dictates the line code used for transmission as can be seen below:

Bit Rate (Mbits)	Number of 64 Kbit Channels	Permitted clock deviation (ppm)	Interface code	Preferred medium/line code		
				Balanc ed	Coaxial	Optical Fiber
2.048	30	±50	AMI	HDB3		
8.448	120	±30	HDB3	HDB3	HDB3	
34.368	480	±20	HDB3	HDB3	4B3T 2B1Q	5B6B
139.264	1920	±15	CMI		4B3T	5B6B

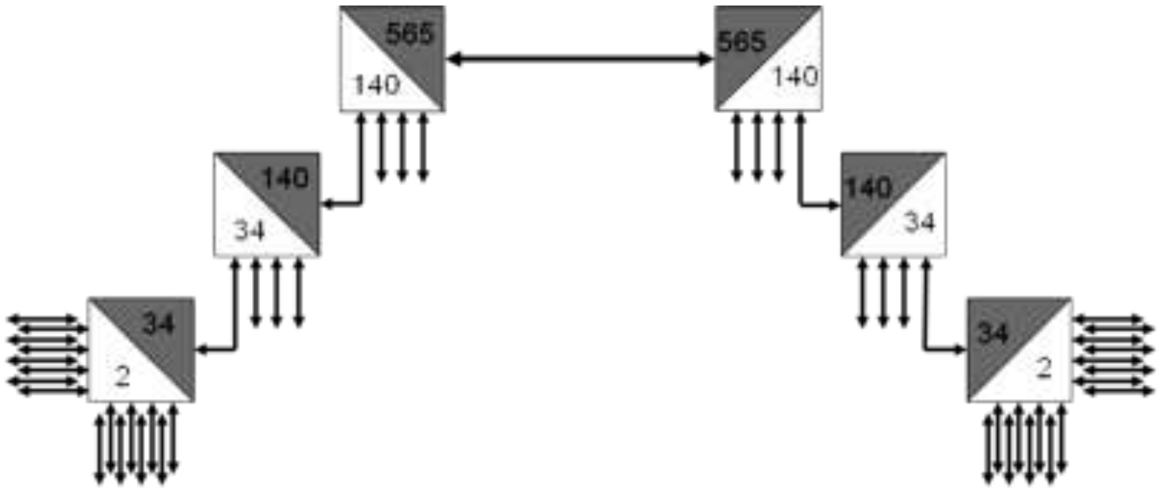
5. NGN – PLESIOCHRONOUS DIGITAL HIERARCHY

Properties of PDH:

- Plesiochronous – "Almost Synchronous."
- Multiplexing of 2 Mbit/s signals into higher order multiplexed signals.
- Laying cable between switch sites is very expensive.
- Increasing traffic capacity of a cable by increasing bit rate.
- 4 lower order signals multiplexed into single higher order signal at each level.



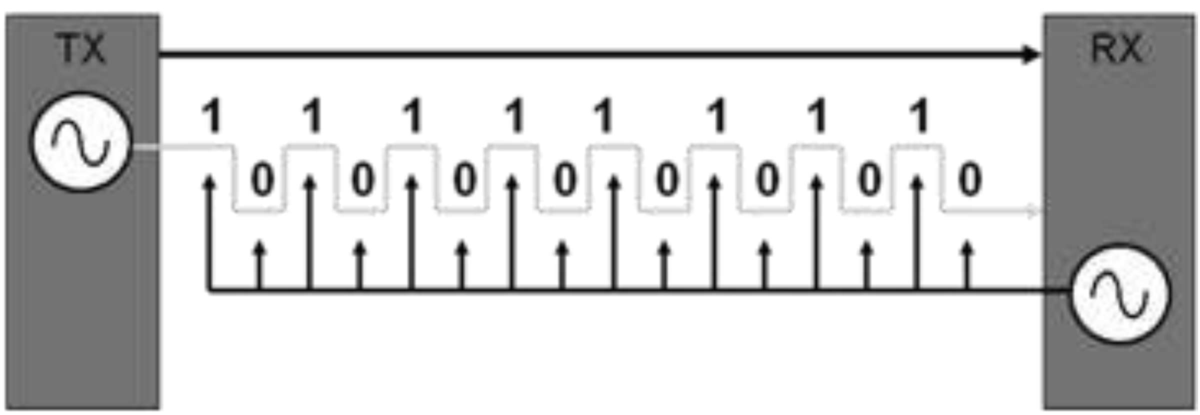
PDH technology allow successive multiplexing of a signal from 2 M – 8 M, from 8 M – 34 M, from 34 M – 140 M and finally 140 M – 565 M systems.



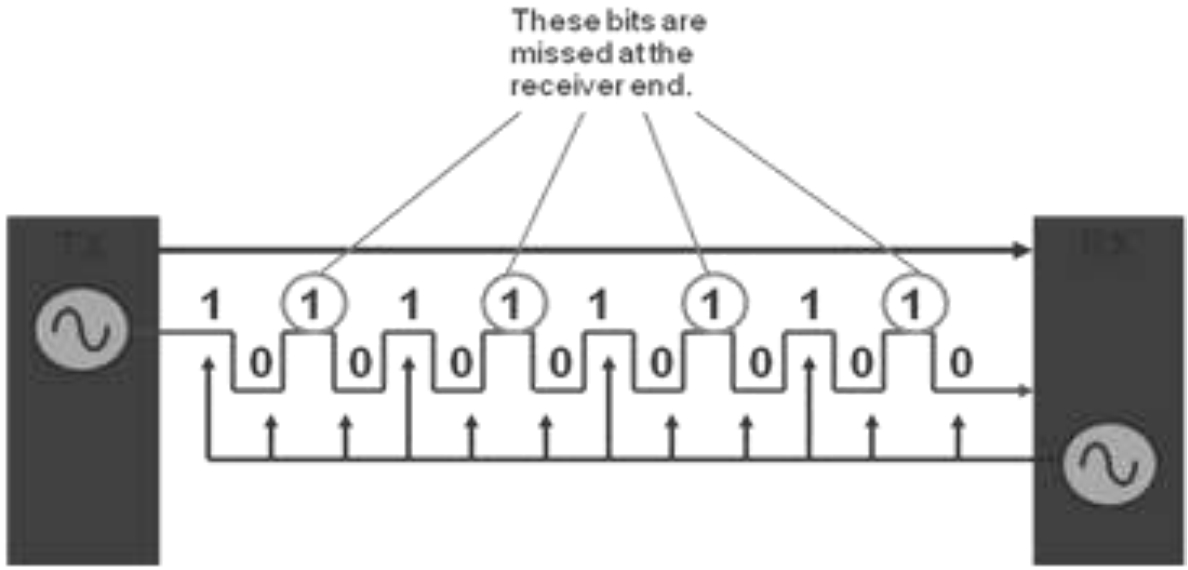
There also existed “jump” or “skip” muxes that would allow multiplexing of 16 2 M signals into a 34 M signal without the intermediate 8 M level.

PDH Limitations

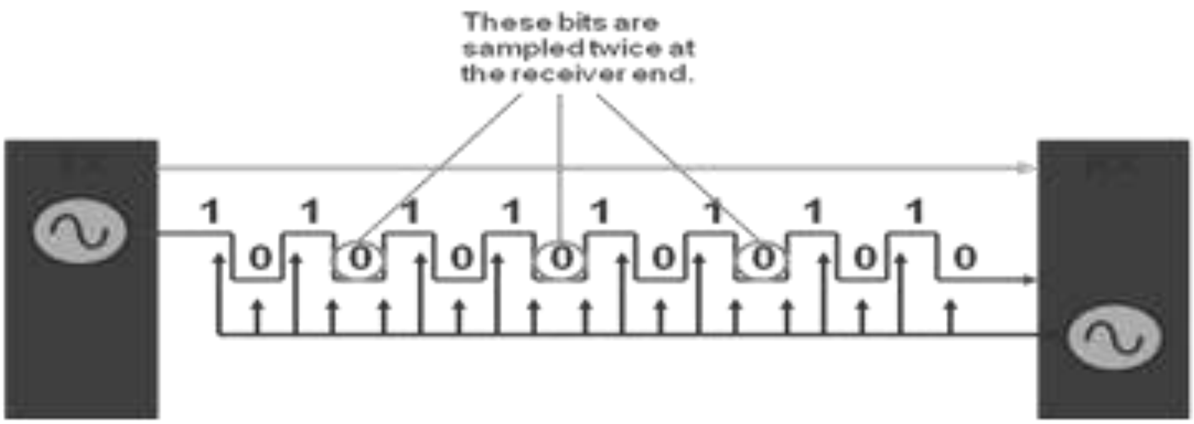
Synchronization: The data is transmitted at regular intervals. With timing derived from the transmitter’s oscillator, the data is sampled at the same rate as it is being transmitted.



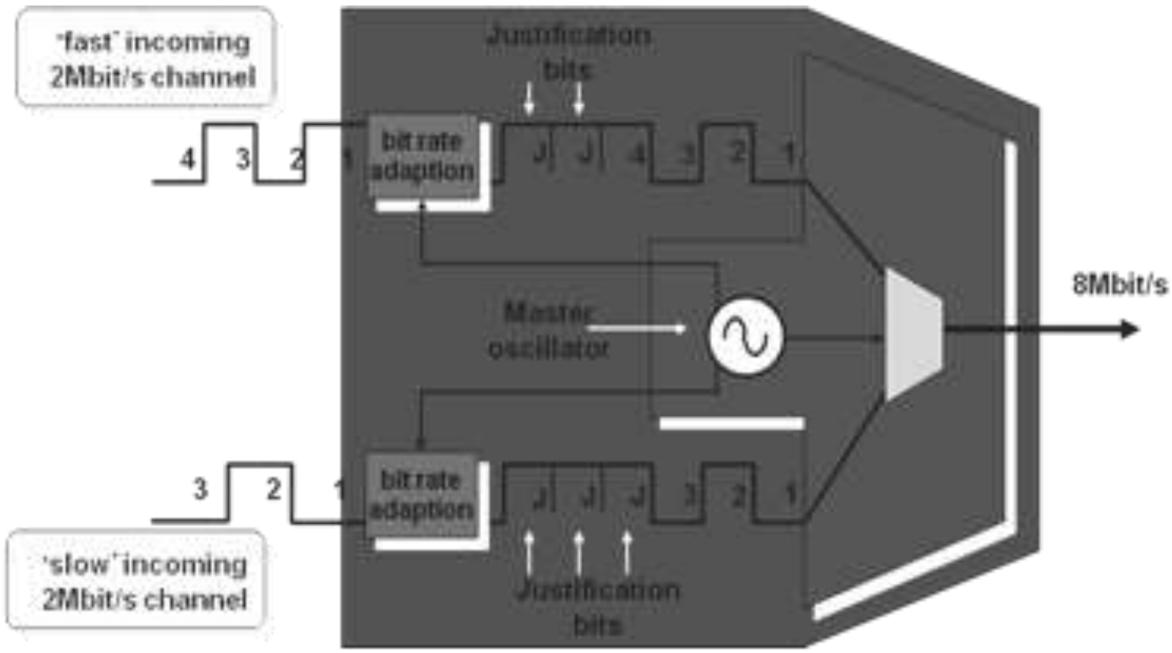
The data is transmitted at regular intervals. With timing derived from the transmitter’s oscillator, the data is sampled at a slower rate than the transmitter. One of the disadvantages of PDH was that each element was synchronized independently. For data to be received correctly, the sampling rate at the receiver end must be the same as the transmission rate at the transmitter end.



The data is transmitted at regular intervals. With timing derived from the transmitter's oscillator, the data is sampled at a faster rate than the transmitter. If the oscillator at the receiver end was running slower than that at the transmitter end the receiver would miss some of the bits of the transmitted signal.



Or, if the receiver clock was running faster than that of the transmitter, the receiver would sample some of the bits twice.



Justification bits are added to lower order signals so that they can be multiplexed at a single rate. The equipment oscillator is used as a timing source for the bit rate adaption process on the lower order and also on the multiplex proceed. Justification bits are discarded at the received end when the signals are de-multiplexed.

Because of the synchronization methods that were used, it was impossible to de-multiplex from a high order signal to the lowest order tributary signal in one piece of equipment. It was necessary to de-multiplex at all levels to access the signal that was being dropped at a site and then re-multiplex all the other channels back up to the higher rate. This meant that there had to be a lot of equipment on the site to accomplish this. This is known as the **PDH Mux Mountain**. All this equipment took up a lot of space on the site and also increased the need for spares to be held on sites.

Lack of resilience in PDH networks meant that if a fiber break occurred, the traffic would be lost. PDH network management simply reports alarms to NOC operators. No diagnosis or remedial tools are available to NOC staff. A maintenance engineer need to be sent on the site with a minimum amount of information. Each network element requires a connection to the DCN network as no facilities exist to carry management information across the PDH network.

Lack of standards for interconnection meant that it was not possible to interconnect equipment from multiple vendors. Equipment could operate on different wavelength, use different bit-rates, or proprietary optical interfaces.