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Ministry of Higher Education and Scientific Research
Badji Mokhtar Annaba University — Université Badji Mokhtar Annaba

Faculty of Technology — Faculté de Technologie
Department of Electronics — Département d'Électronique

L E C T U R E N O T E S

M O D U L E

TELEPHONY

From Analog Switching to 5G and Beyond

COURSE INFORMATION

Specialty	Telecommunications
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MODULE CONTENTS

- **Chapter 1 —** Analog Switching Telephony: History, Evolution, and Architecture
- **Chapter 2 —** Transmission Media: Evaluation Criteria, Copper, Wireless, and Fiber Optics
- **Chapter 3 —** Digital Cellular GSM Telephony: Architecture, Protocols, and Measurement
- **Chapter 4 —** New Generations: 3G / UMTS, 3.5G / HSPA, 4G / LTE, and 5G NR
- **Chapter 5 —** Telephony Interconnection Equipment: Switches, Routers, Gateways, and Interfaces

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ABSTRACT

This lecture note series constitutes a comprehensive academic resource for Third Year Licence students in Telecommunications at Badji Mokhtar Annaba University, covering the ***Telephony*** module in its entirety over five progressive chapters.

Chapter 1 establishes the historical and technical foundations of telephony, tracing the evolution from Bell's point-to-point analog system through Strowger step-by-step and crossbar electromechanical switches to fully electronic stored-program control exchanges. The fundamental principles of **circuit switching** — its three phases of connection establishment, voice transfer, and release — as well as the architecture of the analog Public Switched Telephone Network (PSTN) and its frequency-division multiplexed trunk hierarchy are rigorously examined.

Chapter 2 provides a systematic comparative analysis of the principal **transmission media** employed in telephony networks: twisted-pair copper cable (with its loading coils, characteristic impedance, and DSL bandwidth limitations), coaxial cable, wireless radio links (microwave hops, GEO/LEO satellite systems, and cellular radio with the frequency reuse principle), and single-mode optical fiber (total internal reflection, attenuation windows, chromatic dispersion, EDFA amplification, DWDM, and FTTH/PON access). Each medium is evaluated against a structured set of engineering criteria: bandwidth, attenuation, delay, noise immunity, security, and cost.

Chapter 3 delivers a comprehensive treatment of the **GSM (Global System for Mobile Communications)** second-generation digital cellular standard: its three-subsystem architecture (BSS, NSS, OMS), the detailed functions of every network element (MS/SIM, BTS, BSC, MSC, HLR, VLR, AuC, EIR, OMC), the FDMA/TDMA radio access scheme with its GMSK physical layer, the complete call flow procedures for mobile-originated and mobile-terminated calls, the four types of handover, the three-layer protocol stack (LAPDm/LAPD, RR/MM/CM), the role of SS7 signaling, and the key radio performance indicators (RSSI/RxLev, BER/RxQual, CDR, CSSR) with their measurement methodologies.

Chapter 4 traces the evolution from GSM through the successive **mobile broadband generations**: 3G UMTS with its W-CDMA air interface (spreading codes, RAKE receiver, soft handover, CS+PS core), the 3.5G HSPA/HSPA+ enhancement layer (AMC, HARQ, NodeB scheduling, peak rates to 84 Mbit/s), and 4G LTE with its transformative OFDMA downlink, SC-FDMA uplink, MIMO spatial multiplexing, and all-IP EPC architecture (eNB, MME, S-GW, P-GW). The chapter concludes with an overview of 5G NR (eMBB, URLLC, mMTC, Massive MIMO, flexible numerology, network slicing, and the service-based 5G core architecture) and a forward-looking discussion of 6G research directions.

Chapter 5 examines the **interconnection equipment** that binds heterogeneous networks: digital telephone switches (Time-Space-Time fabric, stored-program control, softswitch architecture), IP routers with VoIP QoS mechanisms (DSCP/EF marking, LLQ, RSVP, jitter buffers), Media Gateways (TDM↔RTP conversion, transcoding, echo cancellation), Signaling Gateways (ISUP↔SIP protocol translation, SIGTRAN/M3UA/SCTP), and the standard physical and protocol interfaces (E1/G.703, T1/ANSI, SIP trunks, Gigabit Ethernet, SDH/OTN).

Keywords

Analog telephony • Circuit switching • GSM • TDMA • FDMA • W-CDMA • OFDMA • MIMO • LTE • 5G NR • Optical fiber • DWDM • SS7 • SIP • Media Gateway • VoIP • QoS • E1/T1 • Network slicing • Massive MIMO

PEOPLE'S DEMOCRATIC REPUBLIC OF ALGERIA
Ministry of Higher Education and Scientific Research

DEPARTMENT OF ELECTRONIC
Third Year Licence (L3) — Academic Year 2021 – 2022

LECTURE NOTES

MODULE:

TELEPHONY

From Analog Switching to 5G and Beyond

Module Structure — Five Chapters

Chapter 1: Analog Switching Telephony: History, Evolution, Principles, and Architecture

Chapter 2: Transmission Media in Telephony: Evaluation Criteria, Electrical Conductors, Wireless, and Fiber Optics

Chapter 3: Digital Cellular GSM Telephony: Networks, Protocols, Architecture, Equipment, and Measurement

Chapter 4: The New Generations of Digital Telephony: 3G/UMTS, 3.5G/HSPA, 4G/LTE, and 5G NR

Chapter 5: Telephony Interconnection Equipment: Switches, Routers, Gateways, and Interfaces

Prepared for L3 Telecommunications Students
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GENERAL INTRODUCTION

I. The Telephone: A Defining Technology of Human Civilization

Few inventions have reshaped the fabric of human society as profoundly and as lastingly as the telephone. From Alexander Graham Bell's first successful voice transmission in 1876 to the 5G smartphones of the 2020s, telephony has been the central pillar of global communication for nearly a century and a half. It has connected families separated by oceans, enabled the coordination of industries and governments, facilitated the conduct of international commerce, and served as the primary emergency communication tool for billions of people in times of crisis.

Yet the telephone system we use today bears little physical resemblance to Bell's original device. Beneath the seamless experience of making a phone call — lifting a handset, dialling a number, speaking with someone thousands of kilometres away in real time — lies an extraordinarily complex engineering infrastructure: kilometres of copper wire and optical fibre, towers bristling with radio antennas, massive server halls housing intelligent switching software, underwater cables spanning oceans, and satellite constellations orbiting the Earth. The invisible machinery of telephony is one of the largest and most sophisticated engineering systems ever built.

Understanding this machinery — how it was conceived, how it evolved, how it works today, and where it is heading — is the purpose of this course. As future telecommunications engineers, you will be called upon not merely to use these systems, but to design, deploy, operate, optimise, and transform them. The knowledge and analytical frameworks developed in this module are therefore foundational to your professional practice.

II. From Analog Circuits to Digital Intelligence: A Century of Transformation

The history of telephony is, at its core, the history of a technology in continuous and accelerating evolution. This evolution can be organized into three major epochs, each marked by a fundamental change in the nature of the communication system:

The Analog Era (1876–1970s): The first hundred years of telephony were characterized by **analog circuit switching**: the creation of a dedicated, continuous electrical path between any two subscribers through a network of mechanical switches. Voice signals were transmitted as

continuously varying electrical waveforms, without any digital encoding or compression. The infrastructure was physical and tangible: copper wire pairs stretching from every home and office to local telephone exchanges, electromechanical Strowger step-by-step switches, crossbar matrices, and coaxial cable bundles carrying hundreds of simultaneous conversations through frequency-division multiplexing. This era established the fundamental concepts of circuit switching, the local loop, the telephone exchange hierarchy, and the national telephone numbering plan — concepts that remain relevant even in today's IP-based systems.

The Digital Revolution (1970s–1990s): The introduction of **Pulse Code Modulation (PCM)** and digital switching in the 1970s initiated the most transformative change in the history of telephony since the invention of the telephone itself. By converting analog voice signals into streams of binary digits — sampled at 8000 times per second, quantized to 8 bits per sample, yielding 64 kbit/s per voice channel — engineers unlocked the full power of digital electronics: near-perfect signal regeneration at repeaters (eliminating the noise accumulation that degraded analog long-distance calls), time-division multiplexing of many channels on a single cable, stored-program control of switching systems (enabling flexible feature provisioning), and eventually, the possibility of carrying voice over the same packet-switched networks used for data. Simultaneously, the first cellular mobile telephone networks emerged (1G analog in the early 1980s, 2G digital GSM in 1991), extending the reach of telephony to mobile subscribers and eventually to every human being on the planet.

The IP Convergence Era (2000s–present): The third and ongoing epoch is the convergence of telephony with the Internet Protocol (IP). Driven by the explosive growth of the internet, the economics of packet networking, and the insatiable demand for mobile broadband data, the telephone network has progressively migrated from circuit-switched TDM infrastructure to **IP-based packet switching**: Voice over IP (VoIP), SIP signaling, LTE's all-IP Evolved Packet Core, and 5G's cloud-native service-based architecture. The distinction between **telephony** and **data networking** — once absolute — has been progressively erased. Today, a phone call is just another IP data stream, differentiated from a web page or a video stream only by its latency, jitter, and reliability requirements.

III. Objectives of this Module

This lecture note series has been designed to provide Third Year Licence students in Telecommunications with a comprehensive, technically rigorous, and practically oriented understanding of telephony systems — from their analog origins to the current 5G generation and beyond. The module pursues five complementary objectives:

1. **Historical and Conceptual Foundation:** To equip students with a thorough understanding of the historical development of telephony, the engineering problems that each generation of technology sought to solve, and the conceptual principles (circuit switching, time-division multiplexing, frequency reuse, packet switching) that underpin all telecommunications networks.
2. **Technical Depth:** To develop the ability to analyze and explain the detailed technical operation of real telecommunications systems — GSM frame structures, W-CDMA spreading codes, OFDMA resource grids, TST switching fabric architectures, SS7 protocol stacks — with the mathematical and engineering precision expected of a graduating telecommunications engineer.
3. **Architectural Understanding:** To build a clear mental model of the architecture of each network generation (PSTN, GSM/2G, UMTS/3G, LTE/4G, 5G NR), the role of each network element, the interfaces between them, and the rationale for each architectural choice.
4. **Systems Integration Perspective:** To understand how heterogeneous networks — analog, digital, mobile, IP — are interconnected through specialized equipment (digital switches, media gateways, signaling gateways, SBCs, routers) and standardized interfaces (E1/T1, SS7/ISUP, SIP, SIGTRAN, GbE), enabling seamless end-to-end communication across technology boundaries.
5. **Measurement and Quality Assessment:** To develop practical skills in evaluating network quality through key performance indicators (RSSI, BER, CDR, CSSR, MOS), understanding the measurement methodologies (drive testing, OMC statistics, protocol analysis), and diagnosing network impairments (coverage holes, interference, congestion, VoIP quality degradation).

IV. Module Structure and Chapter Organization

This module is organized into five chapters that follow the historical and technological evolution of telephony from its analog origins to the current 5G era, with a final chapter dedicated to the interconnection equipment that binds all these network generations together:

Chapter	Title	Core Topics Covered
1	Analog Switching Telephony	Bell's invention; point-to-point limitations; Strowger and crossbar switches; circuit switching (3 phases); DTMF; local loop; MDF; exchange hierarchy; FDM multiplexing
2	Transmission Media	Evaluation criteria (bandwidth, attenuation, security, cost); UTP/coax; radio propagation; microwave/satellite/cellular links; optical fiber (TIR, SMF vs. MMF); EDFA; DWDM; FTTH/PON
3	GSM Digital Cellular	BSS (MS/BTS/BSC) + NSS (MSC/HLR/VLR/AuC/EIR) + OMS; FDMA/TDMA; frequency reuse; logical channels; MOC/MTC call flows; handover; GMSK; LAPDm; SS7; RSSI/BER/CDR/CSSR KPIs
4	New Generations: 3G to 5G	GSM limitations; UMTS W-CDMA (NodeB/RNC/CS+PS core); HSPA/HSPA+ (AMC/HARQ); LTE OFDMA/MIMO/EPC; LTE-A carrier aggregation; 5G NR (eMBB/URLLC/mMTC, SBA, network slicing, Massive MIMO)
5	Interconnection Equipment	Digital switches (TST/STS, SPC, softswitch); IP routers + QoS (DSCP/EF, LLQ, RSVP, jitter buffer); Media Gateways; Signaling Gateways (ISUP↔SIP, SIGTRAN); E1/T1, SIP trunks, GbE, SDH/OTN interfaces

Each chapter opens with clearly stated **pedagogical objectives** (what the student will be able to do after completing the chapter), develops the technical content through rigorous explanations supported by **principle diagrams, comparison tables, and annotated examples**, and closes with a **comprehensive summary table** of key concepts and terminology. Chapters are designed to be self-contained yet progressively cumulative: each chapter builds on the conceptual and technical foundations established by its predecessors.

V. Prerequisites and Assumed Knowledge

These lecture notes assume that students have successfully completed the following L1 and L2 modules in the Telecommunications Engineering programme:

- **Fundamentals of Electricity and Electronics:** Ohm's law, RC/RL circuit analysis, Fourier series, transfer functions, dBm/dB notation, impedance matching. Essential for understanding signal attenuation, transmission line theory (Chapter 2), and audio filter design.
- **Signals and Systems:** Fourier transform, sampling theorem (Nyquist), modulation theory (AM, FM, PSK, QAM), signal-to-noise ratio. Essential for understanding modulation schemes (GMSK, OFDMA, W-CDMA) in Chapters 3–4.
- **Digital Communications:** Binary coding, channel coding (convolutional codes, Viterbi decoding), BER vs. E_b/N_0 curves, link budget analysis. Essential for Chapters 3–4 (GSM channel coding, HSDPA AMC).
- **Computer Networks and Protocols:** OSI model, TCP/IP stack, IP addressing, routing fundamentals (OSPF, BGP), Ethernet. Essential for Chapter 4 (LTE EPC, 5G SBA) and Chapter 5 (VoIP, SIP trunks, QoS, routers).
- **Electromagnetic Wave Propagation:** Maxwell's equations (qualitative), plane wave propagation, reflection/refraction, antenna gain, Friis transmission equation. Essential for Chapter 2 (wireless propagation, Fresnel zones) and Chapter 3/4 (cell planning, link budgets).

VI. Relationship to Professional Engineering Practice

The content of this module maps directly to the core competency domains of a professional telecommunications engineer, as defined by industry frameworks (ITU-T, ETSI, 3GPP, IEEE) and employer skill requirements:

- **Network Design and Planning:** Cell planning (Chapter 3), coverage and capacity dimensioning, frequency allocation, interference analysis, link budget calculation.
- **Network Operation and Maintenance:** KPI monitoring and analysis (Chapter 3), fault isolation, drive testing, OMC-R statistics interpretation, congestion management.
- **Network Engineering and Rollout:** Technology migration (2G→3G→4G→5G, Chapter 4), interconnection equipment selection (Chapter 5), interface provisioning, gateway configuration.
- **Systems Integration:** PSTN-VoIP interworking (Chapter 5), SS7-SIP gateway deployment, E1/SIP trunk provisioning, QoS policy design.
- **Research and Innovation:** Understanding the engineering principles behind 5G (Chapter 4) and emerging 6G technologies provides the conceptual foundation for graduate research in wireless communications, network architecture, and signal processing.

VII. How to Use These Lecture Notes

These notes have been written to serve multiple complementary functions during your studies:

- **Primary study reference:** Each chapter provides a complete and self-consistent treatment of its topic. Read each chapter sequentially before the corresponding lecture, using the pedagogical objectives to orient your reading, and the summary table at the end to consolidate your understanding.
- **Technical reference during exercises:** The comparison tables, formula boxes, and principle diagrams (TDMA/FDMA grid, TST architecture, E1 frame structure, LTE resource grid) are designed to serve as quick-reference materials during problem-solving sessions and laboratory exercises.
- **Examination preparation:** The summary tables at the end of each chapter enumerate the key concepts, definitions, and quantitative relationships that constitute the minimum required knowledge for examinations. Students should ensure they can not only define each term but explain it in their own words and apply it to numerical or analytical problems.
- **Bridge to advanced study:** Each chapter's concluding "Looking Ahead" paragraph explicitly links the chapter's content to more advanced topics in graduate-level courses (advanced wireless communications, network management, optical communications systems, network security) to contextualise the material within the broader curriculum.

A Note on Standards References

Throughout these notes, references are made to international standards documents from ITU-T, 3GPP, IETF, ANSI, and ETSI. While the full texts of these standards are not required reading for this module, students are strongly encouraged to consult the relevant documents (most of which are freely available online from the respective standards bodies' websites) when seeking deeper technical detail on any specific topic. The bibliography at the end of this volume provides the key references.

DEPARTMENT OF ELECTRONIC

Third Year Licence (L3) | Module: Telephony

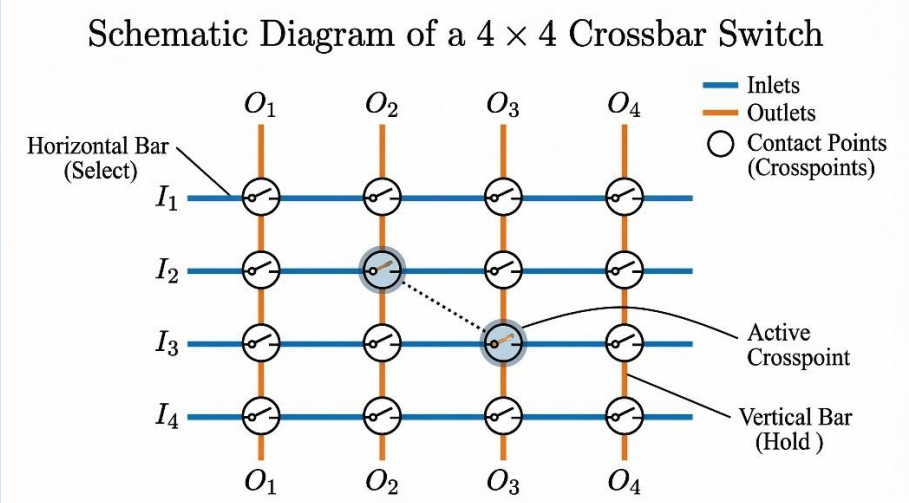
CHAPTER 1

Analog Switching Telephony:

History, Evolution, Principles, and Architecture



(1) A historical photograph of manual telephone exchange operators at a switchboard, circa 1920s.



(2) A labeled schematic diagram of a crossbar switch matrix showing vertical and horizontal conductors with contact points at each intersection. These images will anchor the student's mental model before the technical content begins.

1. Introduction to Chapter 1

1.1 Pedagogical Objectives

Upon successful completion of this chapter, the student will be able to:

- **Define** the concept of analog switching telephony and explain its foundational role in modern telecommunications.
- **Trace** the historical evolution of telephony from the earliest point-to-point connections to fully automated electromechanical switching systems.
- **Explain** the fundamental principle of circuit switching, including the three operational phases: connection establishment, data transfer, and connection release.
- **Describe** the mechanisms of analog voice transmission, including bandwidth constraints, signal degradation, and the implications for long-distance communication.
- **Identify** the principal components and hierarchical architecture of an analog Public Switched Telephone Network (PSTN).
- **Distinguish** between Pulse Dialing and Dual-Tone Multi-Frequency (DTMF) signaling, and explain their technical operation.

1.2 Contextual Importance

Analog telephony represents one of the most significant engineering achievements of the nineteenth and twentieth centuries. For over a century, it served as the primary medium for real-time voice communication across cities, nations, and continents. Its infrastructure — the copper wires, the switching exchanges, and the hierarchical network architecture — laid the physical and conceptual groundwork upon which all subsequent generations of communication networks, including digital telephony and the modern Internet, have been built.

Understanding analog switching telephony is therefore not merely an exercise in historical appreciation; it is an essential prerequisite for grasping why modern networks are designed as they are. Concepts such as **circuit switching** versus **packet switching**, **bandwidth allocation**, **multiplexing**, and **network hierarchy** all have their origins in the challenges first encountered and solved by analog telephony engineers. A thorough command of this chapter

will equip the student to engage critically and analytically with all subsequent modules in this course.

2. Historical Overview and Evolution of Analog Telephony

2.1 The Origins: Bell and the Point-to-Point Model (1876–1880s)

The history of electrical voice communication begins with the invention of the telephone, credited to **Alexander Graham Bell** in 1876. Bell's device converted acoustic pressure waves (sound) into varying electrical currents and transmitted them along a conductive wire to a receiving instrument, where the electrical signal was reconverted into sound. This was a revolutionary breakthrough.

The first telephone deployments operated on a simple **point-to-point** model: a single, dedicated wire directly connected two subscribers — typically a business office to its owner's residence, or two floors of a factory. While functional for these limited purposes, this model suffered from a fundamental and insurmountable scalability problem:

> **The Point-to-Point Scalability Problem**
>
> In a network of N subscribers where every subscriber must be directly connected to every other subscriber, the number of required links is:
>
> **$L = N \times (N - 1) / 2$**
>
> For 10 subscribers: L = 45 links. For 1,000 subscribers: L = 499,500 links. For 10,000 subscribers: L = 49,995,000 links. This exponential growth rendered direct point-to-point connectivity completely impractical for any real urban telephone network. The logical solution was the introduction of a central intermediary: the telephone exchange, or switch.

2.2 The Manual Era: The Human Switch (1878–1920s)

The first telephone exchanges opened in New Haven, Connecticut (USA) in 1878 and quickly spread worldwide. These exchanges operated on a deceptively simple but labor-intensive principle: a human operator served as the network's switching element.

Subscribers were physically connected to a central **manual switchboard** via a dedicated copper wire pair (the local loop). When subscriber A wished to speak with subscriber B, the following sequence occurred:

1. Subscriber A lifted the handset (went **off-hook**), causing a physical lamp or signal to illuminate at the operator's console.
2. The operator plugged a **patch cord** into subscriber A's jack and asked, 'Number please?'
3. The operator was informed of the desired destination (subscriber B's number).
4. The operator plugged the other end of the patch cord into subscriber B's jack and activated a **ringing generator** to alert subscriber B.
5. When subscriber B answered, the two parties were physically connected via the patch cord and could converse.
6. When the call concluded, both parties went **on-hook**, and the operator disconnected the patch cord, freeing the circuit.

The operator was therefore the **first switching element** in telecommunications history — a human being who established, supervised, and released dedicated circuit paths between subscribers. This is a conceptually perfect illustration of the **circuit switching paradigm** that we will formalize in Section 3.

> **Analogy for Students**
>
> *Think of the manual telephone exchange as a very large electrical 'patch bay' (similar to those used in recording studios), with a knowledgeable human operator who knows the 'address' (phone number) of every socket and physically creates a wire path between any two requested endpoints upon demand.*

2.3 The Automatic Era: Electromechanical Switching

2.3.1 The Strowger Switch (Step-by-Step, 1889)

The manual operator model, while functional, had obvious limitations: labor cost, limited speed (an operator could only handle one call at a time), lack of privacy (the operator heard the called number), and vulnerability to human error. The first practical solution came

from **Almon B. Strowger**, an American undertaker who, legend has it, believed a competitor's wife (a telephone operator) was diverting his business calls. In 1889, he patented the world's first **automatic telephone exchange** mechanism.

Technical Principle of the Strowger (Step-by-Step) Switch: The Strowger switch is an electromechanical device that operates in two physical dimensions:

- **Vertical movement**: Driven by pulses on the telephone line (corresponding to the dialed digit), a rotating wiping arm steps vertically through **10 levels**, one step per received pulse.
- **Rotary movement**: After reaching the correct vertical level (row), the arm sweeps horizontally across **10 contacts** per row until it finds an available line.

In a typical 2-digit Strowger system, the first dialed digit moves the arm to the correct row (1 through 10), and the second dialed digit selects the contact within that row (1 through 10), thus reaching one of 100 possible destinations. For larger networks, multiple Strowger stages were cascaded (e.g., three stages for a 3-digit selection, supporting 1,000 lines).

The **Pulse Dialing** system (see Section 3.3) was specifically designed to communicate with Strowger switches: each digit was encoded as a corresponding number of short electrical interruptions on the line, which mechanically stepped the switch arm.

Limitations: Strowger switches were inherently **sequential** in their selection process (they could not 'look ahead' for the best path), were mechanically complex, generated considerable acoustic noise, required significant maintenance, and their **step-by-step** architecture made network-level optimization extremely difficult.

2.3.2 The Crossbar Switch (1938 onward)

The **crossbar switch**, developed primarily by Bell Labs in the 1930s and deployed commercially from 1938 onwards, represented a major architectural advance over the Strowger switch. It addressed the key limitations of step-by-step selection by introducing a **matrix-based switching fabric** with a separate, centralized control logic.

Technical Principle of the Crossbar Switch: Imagine a rectangular grid composed of **N horizontal conductors** (representing input lines) and **M vertical conductors** (representing output lines). At every intersection of a horizontal and a vertical conductor, there is an

electromechanical **contact point** (a crosspoint). Under normal conditions, all crosspoints are open (no connection).

To connect input line **i** to output line **j**, the control logic closes **only the single crosspoint** at the intersection of horizontal conductor **i** and vertical conductor **j**. A dedicated electrical path is thus established between the two lines. Multiple simultaneous calls can be active across the matrix simultaneously, provided no two calls share the same horizontal or vertical conductor.

> **Crossbar vs. Strowger: Key Architectural Differences**
>
> Strowger: Distributed control (each switch stage acts independently, sequentially).
>
> Crossbar: Centralized control (a separate 'marker' circuit identifies a free path across the entire matrix and closes the appropriate crosspoints simultaneously).
>
> Strowger: Sequential path selection (step-by-step, cannot backtrack).
>
> Crossbar: Simultaneous path selection (the entire route is found and established at once), faster and more flexible.

2.3.3 Semi-Electronic Switching (1960s–1970s)

As transistors and, later, integrated circuits became available, engineers began incorporating **solid-state electronic components** into the **control plane** of telephone exchanges. These are called **semi-electronic** (or stored-program control, SPC) switches because they used:

- **Electronic (software-based) control:** A dedicated computer running stored-program software replaced the electromechanical logic of the Strowger or Crossbar control circuits. This computer received signaling information, determined the optimal path through the exchange, and issued commands to close and open switching elements. Software control offered enormous flexibility: new features (call forwarding, conferencing, billing) could be added by modifying software rather than rewiring hardware.
- **Electromechanical switching fabric:** The actual electrical path for the voice signal was still established through a physical crossbar or reed-relay matrix. The **analog voice signal** continued to travel along a dedicated copper path — only the control intelligence was electronic.

Semi-electronic switching dominated the world's telephone networks from the late 1960s through the 1980s and achieved very high reliability and call capacity. It represents the final major stage of purely analog switching before the widespread adoption of digital switching (PCM-based) and later VoIP systems.

3. Key Principles of Analog Switching

3.1 Circuit Switching: The Fundamental Paradigm

The defining operational principle of the analog telephone network is **circuit switching**. This concept must be understood with precision, as it stands in stark contrast to the **packet switching** paradigm that underlies modern data networks (the Internet).

> **Definition: Circuit Switching**
> Circuit switching is a communication method in which a dedicated, continuous, exclusive physical (or logical) path is established between two communicating parties through the network BEFORE data transfer begins, is maintained for the ENTIRE duration of the communication session, and is explicitly released AFTER the session ends.

The circuit switching process can be decomposed into three well-defined phases:

Phase 1: Connection Establishment (Call Setup)

When subscriber A initiates a call to subscriber B, a **signaling process** propagates through the network. Each intermediate switching exchange examines the dialed digits, determines the appropriate outgoing trunk (inter-switch link), reserves one channel on that trunk, and passes the signaling request forward. This process continues until subscriber B's local exchange is reached and subscriber B's handset rings.

By the time subscriber B answers, a **complete, uninterrupted electrical path** has been established through every intermediate switch between A and B. All required resources (circuits, crosspoints, trunk channels) have been **exclusively allocated** to this call. No other call may use these resources while the connection is active.

Phase 2: Data Transfer (Voice Transmission)

With the circuit established, the analog voice signal flows continuously along the dedicated path. The signal travels in both directions simultaneously (**full-duplex**) over the same physical wire pair or allocated channel. The network is transparent to the content of the conversation — it merely provides and maintains the electrical conduit.

Crucially, the allocated bandwidth and resources are consumed for the **entire duration** of the call, **regardless of whether speech is actually occurring**. Silent pauses in a conversation continue to consume network resources — a fundamental inefficiency of circuit switching compared to packet switching, which allocates bandwidth only when data is actually present.

Phase 3: Connection Release (Call Teardown)

When one or both parties hang up (go **on-hook**), an **on-hook signal** is detected by the local exchange. A signaling message is propagated through the network, instructing each intermediate switch to **release** (open) the crosspoints and free the trunk channels that were allocated to this call. These resources then become available for new calls.

> **Analogy: Circuit Switching as a Highway System**
>
> *Imagine circuit switching as reserving an exclusive lane on a motorway from Paris to Lyon before beginning your journey. That entire lane is reserved for you alone from departure to arrival. If you stop at a rest area, the lane remains reserved and unavailable to other drivers. When you reach Lyon and release it, other drivers may use that lane. Packet switching (contrast) would be like sharing all lanes with all other drivers, each car carrying a parcel of your goods independently — no exclusive reservation, but much more efficient use of road capacity.*

3.2 Analog Voice Transmission

The human voice produces sound waves — variations in air pressure — across a frequency spectrum that extends from approximately 20 Hz to 20,000 Hz (20 kHz). However, the **intelligibility** of speech (the ability to recognize words and understand the speaker) is largely preserved within a much narrower band of frequencies.

For practical and economic reasons, the analog telephone network was designed to transmit a frequency band of approximately **300 Hz to 3,400 Hz**. This **3.1 kHz bandwidth** was

found to preserve sufficient intelligibility and speaker recognition for telephone communication while minimizing the technical demands on the transmission infrastructure.

The voice signal within this band is a **continuous analog signal**: its amplitude and phase vary smoothly and continuously over time. In the electrical domain, the telephone's microphone (carbon granule or later dynamic transducer) converts acoustic pressure variations into **proportional voltage variations** on the local loop. These voltage variations are amplified and transmitted along the copper wire pairs and through the switching matrix.

Signal Degradation on Long-Distance Circuits: Copper wires introduce **resistance**, which causes signal amplitude to decrease (attenuation) with distance. They also introduce **capacitance** between the conductors, which tends to suppress higher-frequency components more than lower ones, causing frequency-dependent distortion. On long-distance circuits, **amplifiers** (repeaters) were installed at regular intervals to compensate for attenuation. However, each amplification stage also amplified **thermal noise** present on the circuit, leading to a progressive degradation of the signal-to-noise ratio (SNR) over very long distances. This was a fundamental limitation of analog transmission, later resolved by the transition to **digital transmission** using Pulse Code Modulation (PCM).

3.3 Dialing and Signaling

For a telephone call to be completed automatically (without a human operator), the subscriber must be able to communicate the **destination telephone number** to the switching network. This is the function of the **dialing and signaling system**. We distinguish two classical dialing methods and several categories of line signaling.

3.3.1 Pulse Dialing (Decimal Dialing / Loop Disconnect)

Pulse dialing, also known as **loop disconnect** or **decadic dialing**, was the original dialing method designed specifically for Strowger automatic exchanges. Its technical operation is as follows:

- When the subscriber's dial is turned to a digit and released, the **dial mechanism** opens and closes the DC loop (the current-carrying circuit) at a rate of **10 pulses per second** (in the European standard; 20 pps was used in some systems).

- The number of loop interruptions corresponds exactly to the dialed digit. For example, dialing the digit '5' produces 5 brief interruptions of the DC current; dialing '0' produces 10 interruptions.
- Each interruption typically has a duration of approximately **60 ms off** followed by **40 ms on** (a 60/40 make-break ratio, though this varied by country).
- At the exchange, the **line equipment detects these current pulses** and steps the Strowger selector mechanism accordingly, or (in later systems) a register counts the pulses and stores the digit for subsequent processing by the central control.

Pulse dialing was robust (requiring only simple DC loop switching) but was slow (a digit '9' took approximately 900 ms to transmit) and was inherently mechanical, limiting its accuracy in poor line conditions.

3.3.2 Dual-Tone Multi-Frequency Signaling (DTMF)

DTMF, standardized by Bell Labs under the name **Touch-Tone** and standardized internationally by the ITU-T, was developed in the 1960s as a faster, more reliable alternative to pulse dialing. Its principle is:

- The telephone keypad is organized in a **4×3 matrix** (for a standard keypad: rows = 4 frequency groups, columns = 3 frequency groups). Each key corresponds to a unique pair of sinusoidal tones: one from the **low-frequency group** (697, 770, 852, or 941 Hz) and one from the **high-frequency group** (1209, 1336, or 1477 Hz).
- Pressing a key causes the telephone's electronics to **simultaneously transmit two pure sinusoidal signals** at the corresponding frequencies. For example, pressing digit '5' transmits 770 Hz + 1336 Hz simultaneously.
- At the exchange, a **DTMF receiver** (a bank of narrow-band filters or digital signal processing algorithms) detects the presence and frequency of the two simultaneous tones and decodes the corresponding digit.

DTMF signaling is transmitted **in-band** (within the same 300–3400 Hz voice channel), making it compatible with any existing telephone connection. A single DTMF digit requires only **50 ms** minimum tone duration, making digit transmission approximately 18 times faster

than pulse dialing. DTMF became the global standard for subscriber signaling and remains in use today for voicemail navigation and interactive voice response (IVR) systems.

DTMF Frequency Assignment Table

	1209 Hz	1336 Hz	1477 Hz
697 Hz	1	2	3
770 Hz	4	5	6
852 Hz	7	8	9
941 Hz	*	0	#

3.3.3 Line Signaling States

Beyond dialing, the telephone system relies on a set of **line signaling conventions** to communicate the status of the telephone and the call to the exchange:

- **On-hook:** The handset rests in the cradle. The local loop is open (no current flows). The subscriber is idle and available to receive calls.
- **Dial Tone:** When the subscriber lifts the handset (goes off-hook), the local exchange detects current flow on the loop and responds by applying a **dial tone** (in most countries: 350 Hz + 440 Hz, continuous) to indicate that the exchange is ready to receive dialed digits.
- **Ringing:** The calling exchange applies a high-voltage AC ringing signal (typically 75–90 V AC at 25 Hz) to the called subscriber's loop to activate the telephone bell or electronic ringer. Simultaneously, a **ringback tone** is applied to the calling party's line (a different tone, typically 400 Hz + 450 Hz, ringing cadence) to indicate that the distant telephone is being rung.
- **Busy Tone:** If the called line is engaged (off-hook) or no free path exists through the network, a **busy tone** (typically 400 Hz, interrupted at 0.5 s on / 0.5 s off cadence) is applied to the calling party's line.
- **Off-hook:** The handset is lifted. Current flows on the DC loop. The subscriber is either initiating a call, receiving a call, or actively conversing.

4. Architecture of an Analog Telephone Network

The analog Public Switched Telephone Network (PSTN) can be analyzed as a layered architectural system, proceeding from the individual subscriber's terminal equipment through the local connection infrastructure, through the hierarchy of switching exchanges, to the long-distance trunk network. Each layer performs a distinct and essential function.

4.1 The Subscriber Terminal: The Analog Telephone Set

The **analog telephone set** (poste téléphonique) is the subscriber's interface to the telephone network. Its principal functional blocks are:

- **The Transmitter (Microphone):** Converts acoustic speech energy (sound waves) into a proportional analog electrical signal. Early telephones used **carbon granule microphones**, in which the compression of carbon particles by sound waves varied the resistance (and thus current) in the circuit. Modern analog telephones use **electret condenser** or dynamic microphones.
- **The Receiver (Earphone):** Converts the received electrical signal from the network back into acoustic sound via an electromagnetic coil acting on a diaphragm.
- **The Hybrid Circuit (2-wire / 4-wire conversion):** The local loop uses a single **2-wire pair** (one pair of conductors) to carry both the transmitted and received signals simultaneously. The hybrid circuit separates these two directions of transmission while preventing the subscriber from hearing their own transmitted signal excessively (this residual effect is called **sidetone**).
- **The Ringer:** Detects the incoming ringing signal and activates an audible alert.
- **The Dial (DTMF Keypad or Pulse Dial):** Provides the mechanism for digit entry, as described in Section 3.3.
- **The Hook Switch:** A mechanical switch controlled by the weight of the handset, providing the on-hook / off-hook signaling function.

4.2 The Local Loop (Subscriber Line)

The **local loop** (also called the **subscriber line** or, in French, **la boucle locale**) is the physical transmission medium connecting the subscriber's telephone set to the nearest telephone exchange (the local exchange or central office). In the analog PSTN, the local loop consists of a single **unshielded twisted pair (UTP) of copper conductors**

Why a twisted pair? Twisting the two conductors of the pair at regular intervals causes electromagnetic interference induced on the pair to cancel out between the two conductors (common-mode rejection). This significantly reduces susceptibility to external noise sources (nearby electrical equipment, power lines).

The local loop carries two physically distinct functions over the same wire pair:

- **DC feeding:** The local exchange applies a **direct current** (typically 48 V battery, negative polarity at the exchange) along the local loop. This current powers the telephone set's circuitry (no separate mains power supply is required) and serves as the **signaling carrier** for on-hook/off-hook detection and pulse dialing.
- **AC voice signal:** The analog voice signal (300–3400 Hz) is superimposed on the DC feeding current as a small-amplitude AC variation.

Electrical Constraints of the Local Loop: The local loop is characterized by its **resistance per unit length** (typically 40–100 Ω/km for the conductor gauges used). The total loop resistance determines whether the DC current is sufficient to correctly feed the telephone and operate the line signaling circuits. A maximum loop resistance limit was traditionally set (e.g., 1800 Ω for POTS — Plain Old Telephone Service), defining a maximum achievable distance of approximately 5–7 km without additional loading or amplification.

4.2.1 The Main Distribution Frame (MDF)

At the entrance to the telephone exchange building, all incoming subscriber local loop pairs are terminated at the **Main Distribution Frame (MDF)** (in French: **Répartiteur Principal**). The MDF is a large, structured wiring frame performing the following functions:

- **Physical termination:** Each individual copper wire pair from every subscriber's local loop is physically terminated and labeled at the MDF.
- **Surge protection:** Protector modules on the MDF guard the sensitive exchange equipment against high-voltage transients arriving from the subscriber network (lightning strikes, power line contact).
- **Cross-connection:** The MDF allows any subscriber's local loop pair to be cross-connected ("jumpered") to any line card port within the exchange. This provides administrative flexibility for line provisioning, relocation, and testing without requiring physical rewiring within the exchange.
- **Testing access:** The MDF provides a controlled access point for connecting test equipment to any subscriber's local loop without entering the subscriber's premises.

4.3 The Telephone Exchange (Switching Office)

The **telephone exchange** (central téléphonique, or central office in North American terminology) is the core network element of the analog PSTN. Its primary functions are:

- **Concentration:** A large number of subscriber local loops (potentially thousands) are concentrated onto a smaller number of internal switching resources and inter-exchange trunks. This is based on the statistical observation that the **traffic intensity** on any given subscriber line is low (most subscribers are idle most of the time). The **concentration ratio** (e.g., 8:1 or 10:1) is a fundamental economic design parameter.
- **Switching (Commutation):** The exchange establishes dedicated circuit paths between any two subscriber lines or between a subscriber line and an inter-exchange trunk, as directed by the received dialed digits and the control system.
- **Distribution (Expansion):** Conversely, the exchange distributes incoming calls from inter-exchange trunks to the appropriate subscriber local loops.
- **Signaling generation:** The exchange generates and detects all supervisory tones (dial tone, busy tone, ringing signal, ringback).
- **Billing (Call Data Recording):** The exchange records details of each completed call (calling number, called number, duration, start time) for billing purposes.

4.3.1 Exchange Hierarchy in the Analog PSTN

A national analog telephone network is not a flat structure in which all exchanges are directly interconnected. Rather, it is organized into a **hierarchical pyramid** of exchange classes, each serving a different geographic scope and traffic level:

Exchange Level	Designation	Geographic Scope	Connected To
Class 5 (Local)	Local Exchange / Central Office	Neighborhood / City District	Subscriber local loops; Tandem exchanges
Class 4 (Tandem)	Tandem / Junction Exchange	City / Region	Multiple local exchanges; Toll exchanges
Class 3 (Primary Toll)	Primary Toll Center	Province / State	Tandem exchanges; Secondary toll centers
Class 2 (Secondary Toll)	Secondary Toll Center	National Region	Primary toll centers; International gateways
Class 1 (International)	International Gateway / Transit Exchange	International	National top-level exchanges; Foreign network gateways

This hierarchical structure minimizes the total number of inter-exchange trunks required (compared to a fully meshed flat network), while ensuring that any call can be completed by routing upward through the hierarchy until a common ancestor exchange is found, and then routing downward to the destination.

4.4 Inter-Exchange Trunks and Frequency Division Multiplexing (FDM)

Directly connecting every pair of exchanges at each hierarchical level with a separate physical wire pair for each possible simultaneous call would be economically and physically impractical. Two solutions were employed:

4.4.1 Trunk Groups

Rather than provisioning dedicated subscriber-grade wire pairs, exchanges are interconnected by **trunk groups**: bundles of high-capacity transmission links that are shared by many simultaneous calls through **statistical multiplexing** (engineered using traffic theory, specifically the Erlang B formula, to size the trunk group for acceptable call blocking probability).

4.4.2 Frequency Division Multiplexing (FDM)

On long-distance transmission media (coaxial cables, early radio relay systems, submarine cables), **Frequency Division Multiplexing (FDM)** was employed to transmit many analog telephone conversations simultaneously over a single physical medium. The principle is:

- **Channel separation:** Each individual 4 kHz voice channel (300–3400 Hz bandwidth, plus guardbands) is assigned a unique **frequency slot** within the available transmission bandwidth of the medium.
- **Modulation:** Each voice signal **modulates a distinct carrier frequency** using **Single Sideband Amplitude Modulation (SSB-AM)**. The resulting modulated signals each occupy a 4 kHz band, centered on their respective carrier frequency.
- **Combination:** All modulated channels are combined (summed) into a single composite broadband signal and transmitted simultaneously over the shared medium.
- **Demodulation:** At the receiving exchange, each channel is separated using a bank of **bandpass filters** tuned to the respective carrier frequency, and demodulated back to the original 300–3400 Hz baseband voice signal.

The ITU-T standardized FDM hierarchies for the analog PSTN:

- **Basic Group:** 12 voice channels, occupying 48 kHz (60–108 kHz band).
- **Super Group:** 5 Basic Groups = 60 voice channels, occupying 240 kHz.
- **Master Group:** 5 Super Groups = 300 voice channels.
- **Super Master Group:** 3 Master Groups = 900 voice channels on a single coaxial pair.

FDM was the backbone of international analog telephony from the 1940s through the 1980s, enabling transcontinental and transoceanic telephone communication before being superseded by digital transmission technologies (Pulse Code Modulation, SDH/SONET).

5. Chapter Summary: Key Concepts to Remember

The following table summarizes the essential concepts and terminology introduced in this chapter. Students should ensure they can define, explain, and apply each of these elements.

Concept / Term	Essential Definition / Key Point
Point-to-Point Topology	Direct dedicated link between two subscribers; impractical at scale due to $O(N^2)$ link growth.
Manual Exchange	Human operator as the switching element; establishes and releases circuit paths manually via patch cords.
Strowger Switch	First automatic switch (1889); electromechanical, step-by-step, driven directly by loop-disconnect pulses; distributed control.
Crossbar Switch	Matrix-based switch (1938+); centralized control via 'marker'; simultaneous path selection; more efficient and reliable.
Semi-Electronic Switch	Crossbar-type fabric with stored-program electronic control; combines software flexibility with electromechanical switching.
Circuit Switching	A dedicated, exclusive, continuous physical path established for the full duration of a call; three phases: setup, transfer, teardown.
Voice Bandwidth (PSTN)	300 Hz to 3,400 Hz (3.1 kHz bandwidth); sufficient for speech intelligibility; limits audio quality compared to full audible range.
Pulse Dialing	DC loop interruptions at 10 pps; N interruptions = digit N; designed for Strowger switches; slow but robust.
DTMF	Simultaneous dual-tone in-band signaling; 4 low-freq × 3 high-freq = 12 unique tones; fast (min. 50 ms/digit), universal.
Local Loop	2-wire copper UTP from subscriber to local exchange; carries DC feed (48 V), pulse signaling, and AC voice; limited to ~5–7 km without amplification.
MDF (Main Distribution Frame)	Physical termination point of all local loops at the exchange; provides surge protection, cross-connection, and test access.

Concept / Term	Essential Definition / Key Point
Telephone Exchange Functions	Concentration (many loops → fewer trunks), Switching (path establishment), Distribution (expanding trunks → loops), Signaling, Billing.
Network Hierarchy	Classes 1–5 from international gateways (Class 1) to local exchanges (Class 5); minimizes trunk requirements through hierarchical routing.
FDM (Frequency Division Multiplexing)	Each analog voice channel occupies a unique 4 kHz frequency slot on a shared broadband medium; SSB-AM modulation; ITU-T standardized hierarchy (Group, Super Group, etc.).

Looking Ahead: From Analog to Digital

Chapter 2 will introduce 'Transmission Media in Telephony: Evaluation Criteria, Electrical Conductors, Wireless, Fiber Optics'.

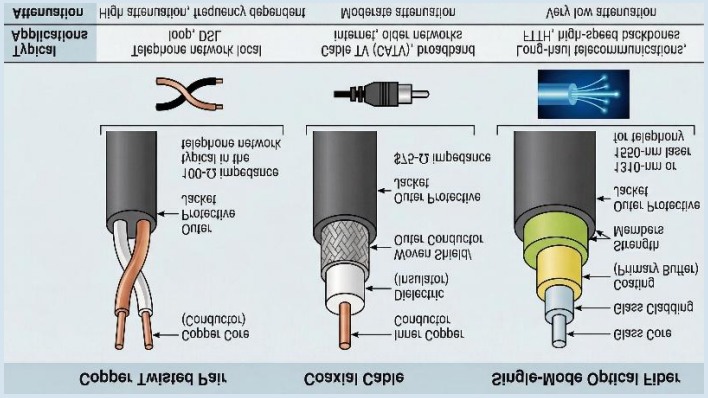
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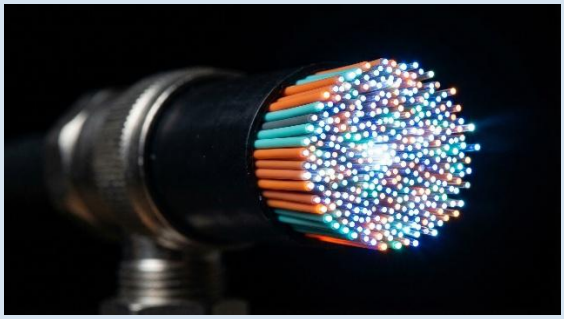
CHAPTER 2

Transmission Media in Telephony:

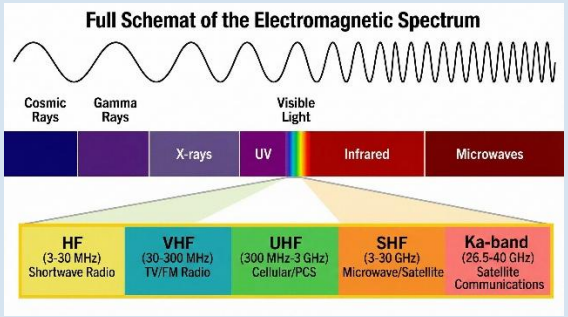
Evaluation Criteria, Electrical Conductors, Wireless Transmission, and Fiber Optics



(1) A cross-section diagram of the three media side-by-side: copper twisted pair, coaxial cable, and single-mode optical fiber, each labelled with key structural components.



(2) A photograph of a fiber-optic cable bundle illuminated at one end.



(3) A schematic of the electromagnetic spectrum with telephony-relevant bands highlighted (HF, VHF, UHF, SHF microwave, Ka-band satellite).

1. Introduction to Chapter 2

1.1 Pedagogical Objectives

Upon successful completion of this chapter, the student will be able to:

- **Enumerate and apply** the principal quantitative and qualitative criteria used to evaluate any transmission medium for telephony and broadband applications.
- **Describe** the physical structure, electrical characteristics, and performance limitations of twisted-pair copper conductors and coaxial cable, and explain their role in the telephone network.
- **Explain** the principal propagation mechanisms of wireless (radio) transmission, including microwave point-to-point links, satellite systems (GEO, LEO), and cellular radio telephony.
- **Describe** the physical principles of optical fiber light guidance (total internal reflection), distinguish between single-mode and multimode fiber, and explain the origins of attenuation and dispersion.
- **Compare** all four major media categories across standardized evaluation criteria and justify media selection for given network deployment scenarios.

1.2 Context and Motivation

Every telephone call — whether from an analog POTS handset, a mobile terminal, or a VoIP client — must be carried at some point by a physical or electromagnetic **transmission medium**. The choice of medium is not incidental; it is one of the most consequential engineering decisions in network design, determining achievable bandwidth, maximum unrepeated distance, susceptibility to noise and interference, infrastructure cost, and ultimately the quality of service experienced by the subscriber.

In the context of the telephone network hierarchy introduced in Chapter 1, transmission media appear at every level: on the **subscriber local loop** (connecting the handset to the local exchange), on **inter-exchange trunk links** (connecting local and tandem exchanges within a city), on **long-distance backbone routes** (connecting cities and countries), and on **international links** (submarine cables, satellite circuits). Each of these environments has

distinct physical constraints, economic requirements, and regulatory frameworks that favor different media solutions.

This chapter provides a rigorous comparative framework — the **evaluation criteria** — and applies it systematically to the three major media families: **electrical conductors** (twisted-pair copper and coaxial cable), **wireless radio systems**, and **optical fiber**. Understanding this framework is essential for analyzing the engineering rationale behind the progressive migration of telephone networks from copper to fiber, and from fixed to wireless infrastructure, over the past five decades.

2. Evaluation Criteria for Transmission Media

Before selecting a transmission medium for a given telephone network segment, the telecommunications engineer must assess the medium against a structured set of performance, economic, and operational criteria. These criteria are universally applied in telecommunications systems engineering and provide a consistent framework for comparative analysis.

2.1 Bandwidth (Frequency Capacity)

Definition: Bandwidth is the range of frequencies that a medium can transmit with acceptable signal fidelity (i.e., without excessive attenuation or distortion). It is expressed in Hertz (Hz) and is the primary determinant of the information-carrying capacity of the medium.

For analog telephony, the minimum required channel bandwidth is **3.1 kHz** (300–3400 Hz). For digital transmission, the achievable **bit rate capacity** is related to bandwidth through Shannon's channel capacity theorem:

> **Shannon's Channel Capacity: $C = B \times \log_2(1 + S/N)$**
>
> Where C = capacity (bits/s), B = bandwidth (Hz), S/N = signal-to-noise ratio (linear). This theorem establishes a fundamental upper bound on throughput, independent of the modulation scheme or coding. A medium with greater bandwidth and/or higher achievable SNR can support proportionally higher information rates.

2.2 Attenuation

Definition: Attenuation is the reduction in signal power as it propagates along the medium, expressed in **decibels per kilometre (dB/km)**. It increases with both distance and signal frequency.

Physical causes of attenuation vary by medium: **resistive losses** (Joule heating in copper), **dielectric losses** (energy absorption in insulating materials), and **Rayleigh scattering and infrared absorption** (in optical glass). High attenuation requires periodic **amplifiers or regenerative repeaters**, increasing infrastructure cost and complexity. The maximum **unrepeated span** of a link is directly governed by the attenuation coefficient and the available signal power margin.

2.3 Propagation Delay

Propagation delay is the time required for a signal to travel from transmitter to receiver. It is determined by the **propagation velocity** of the medium (a fraction of $c \approx 3 \times 10^8$ m/s) and the physical distance:

- **Copper twisted pair:** Propagation velocity ≈ 0.60–0.70 × c. Delay ≈ 5–7 μs/km. Negligible for local loop distances.
- **Optical fiber:** Propagation velocity ≈ 0.67 × c (due to glass refractive index n ≈ 1.5). Delay ≈ 5 μs/km. Measurable but acceptable on intercontinental links.
- **Geostationary (GEO) satellite:** Orbital altitude 35,786 km. One-way propagation delay ≈ 240 ms; round-trip delay ≈ 480–600 ms. **Highly problematic for real-time voice telephony**: causes echo, conversational overlap, and TCP performance degradation. Mandatory ITU-T echo cancellation (G.168) required on all GEO satellite voice circuits.
- **LEO satellite:** Orbital altitude 160–2,000 km. One-way delay ≈ 2–10 ms. Acceptable for voice. Requires large constellations for continuous coverage.

2.4 Noise and Interference Susceptibility

Noise is any unwanted signal superimposed on the desired transmission. The principal metric is the **Signal-to-Noise Ratio (SNR, dB)**. For intelligible telephony, an SNR of at least 20–25 dB is required; below 10 dB, intelligibility fails entirely.

- **Thermal noise:** Present in all electronic components proportional to temperature and bandwidth (Johnson–Nyquist noise). A fundamental physical lower bound; minimized by careful amplifier design.
- **Electromagnetic interference (EMI):** Induced by nearby power lines, motors, lightning, and other electrical equipment. Major concern for copper cables; mitigated by twisting and shielding.
- **Crosstalk:** Mutual electromagnetic coupling between adjacent wire pairs in the same cable sheath. Near-End CrossTalk (NEXT) and Far-End CrossTalk (FEXT) are the dominant forms. A critical limitation for DSL broadband over copper binders.
- **Radio-frequency interference (RFI):** Relevant to both copper conductors (acting as inadvertent antennas) and wireless systems. Wireless systems additionally suffer from **multipath fading** and **co-channel interference** from other radio systems.
- **Optical fiber immunity:** Completely immune to EMI and RFI. The photonic signal does not interact with electromagnetic fields. The dominant noise sources are shot noise and thermal noise at the photodetector receiver, plus Rayleigh scattering within the fiber core. This immunity is a decisive advantage in electrically noisy industrial environments, alongside substations, railways, and high-voltage lines.

2.5 Security

Security refers to the resistance of the medium to unauthorized signal interception.

- **Copper conductors:** Minimal inherent security. Passive inductive tapping (non-invasive) or galvanic tapping (invasive) is technically straightforward and difficult to detect. Any physical access to the cable plant enables interception.
- **Wireless:** The signal propagates into open space and is accessible to any receiver within range. Security depends entirely on **cryptographic encryption** at the protocol layer

(GSM: A5/1 stream cipher; LTE: AES-128; 5G: 256-bit algorithms). Radio-direction-finding can locate transmitters.

- **Optical fiber:** The most inherently secure medium. Light is confined within the glass core; no field is radiated outside the fiber. Tapping requires physically bending or cutting the fiber, causing a measurable drop in received optical power detectable by **Optical Time-Domain Reflectometry (OTDR)**. Exploited in **Quantum Key Distribution (QKD)** systems for provably secure key exchange.

2.6 Cost

Cost must be evaluated across the full lifecycle of the infrastructure:

- **Capital expenditure (CapEx):** Raw material cost, manufacturing, civil works (trenching, conduit for buried cable; tower or pole erection for wireless), terminal equipment (transceivers, amplifiers, optical transmitters/receivers, base station hardware).
- **Operational expenditure (OpEx):** Preventive and corrective maintenance, power consumption (remote amplifiers, active electronics in wireless networks), spectrum licensing fees (wireless only, often very large), and network management.
- **Upgrade/scalability cost:** Adding capacity to an existing fiber pair via DWDM (Dense Wavelength Division Multiplexing) is relatively inexpensive (upgrade terminal equipment only). Increasing copper cable capacity requires laying additional pairs or replacing the entire cable plant — very costly.

2.7 Additional Engineering Criteria

- **Installation ease:** Wireless requires no physical medium between endpoints (major advantage in mountainous terrain, over water, or across protected zones). Fiber requires specialized splicing tools and skills. Copper is the most straightforward to install and terminate.
- **Environmental resilience:** Optical fiber is chemically inert and corrosion-resistant. Copper oxidizes in humid conditions, degrading contact quality and increasing

resistance. Wireless performance degrades with adverse weather (rain, fog, dense foliage) and physical obstructions (buildings, terrain).

- **Maximum unrepeated distance:** Single-mode fiber with EDFA optical amplification: 80–120 km between amplifiers, transoceanic distances achievable. Microwave hop: 50–80 km per hop (line-of-sight). Copper twisted pair: ≈5 km for POTS voice; ≈1 km for high-speed VDSL.

3. Electrical Conductors

Electrical conductors — principally copper-based media — constituted the exclusive physical infrastructure of the telephone network from its inception until the large-scale deployment of fiber optics in the 1980s and 1990s. They remain present in the **subscriber access segment** of many networks today, particularly for xDSL broadband delivery. Two principal forms are covered: the **twisted-pair cable** and the **coaxial cable**.

3.1 Twisted-Pair Copper Cable

3.1.1 Physical Structure and Construction

A twisted pair consists of **two copper conductors** (solid or stranded), each individually insulated (polyethylene or PVC), and then **twisted together at a defined pitch** (lay length, typically 2–20 cm between twists). Multiple such pairs are assembled into a **multi-pair cable** with an outer polyethylene sheath, forming the standard subscriber local loop cable plant.

The physical rationale for twisting: Any electromagnetic field emanating from an external source (power line, motor, lightning) induces voltages on both conductors of a pair. Because the telephone receiver responds only to the **differential voltage** between the two conductors (not the absolute voltage of either), interference that appears equally on both conductors (**common-mode noise**) is rejected. Twisting ensures that over any given length of cable, each conductor is equally exposed to external field sources, so the induced voltages are equal and opposite in each successive half-twist, canceling upon subtraction at the receiver. **Shorter lay length = tighter twist = better common-mode rejection = better EMI immunity.**

3.1.2 Distributed Electrical Parameters

The electrical behavior of a twisted pair is governed by four **distributed parameters** expressed per unit length (per kilometre):

- **Series resistance R (Ω/km):** Ohmic resistance of the copper conductors. Depends on wire gauge (diameter). Typical values for subscriber wire: 100–300 Ω/km for 0.4–0.6 mm diameter. Causes DC voltage drop on the local loop, limiting the range of the 48 V battery feed from the exchange. Also increases with frequency due to the **skin effect** (AC current concentrates near the conductor surface at high frequencies, reducing the effective cross-section).
- **Series inductance L (mH/km):** Magnetic energy stored in the field surrounding the conductor pair. Typically 0.5–1.0 mH/km. Acts to oppose changes in current (inductive reactance ωL increases with frequency).
- **Shunt capacitance C (nF/km):** Electric energy stored between the two conductors separated by the dielectric insulation. Typically 40–100 nF/km. Shunt capacitance **acts as a low-pass filter**: at high frequencies, it provides a low-impedance shunt path to ground, progressively bypassing the signal and causing increasing attenuation with frequency. This is the dominant **bandwidth limitation** of twisted-pair cable.
- **Shunt conductance G (µS/km):** Leakage current through the insulation between the two conductors. Normally very small for high-quality cable; increases significantly with physical insulation damage, moisture ingress, or aging.

Characteristic Impedance of a Transmission Line

$$Z_0 = \sqrt{[(R + j\omega L) / (G + j\omega C)]}$$

At high frequencies (where ωL >> R and ωC >> G), this simplifies to $Z_0 \approx \sqrt{(L/C)}$, a real, frequency-independent value of approximately **100–135 Ω** for standard subscriber wire pairs. Correct impedance termination at both ends minimizes signal reflections (return loss) that would degrade SNR.

3.1.3 Attenuation vs. Frequency

The attenuation of a twisted pair increases with frequency approximately as $\sqrt{f}$ at lower frequencies (where series resistance dominates) and as **f** at higher frequencies (skin effect and dielectric losses). Representative measured values for a standard 0.5 mm diameter copper pair:

Frequency	Attenuation (0.5 mm)	Usable for POTS?	Usable for DSL?
300 Hz	~1 dB/km	Yes	Yes
3.4 kHz	~3 dB/km	Yes	Yes
100 kHz	~7 dB/km	N/A	ADSL only
1 MHz	~20 dB/km	No	VDSL: ≤1 km
17.6 MHz	~70 dB/km	No	VDSL2 only

> ⚠ **Key Engineering Implication**
> The progressive increase of attenuation with frequency means that the usable bandwidth of a copper subscriber loop is **inversely proportional to the loop length**. A 300 m loop (typical in dense urban FTTN deployment) supports VDSL2 at up to 100 Mbit/s. A 5 km loop (typical rural subscriber) supports only ADSL2+ at 1–2 Mbit/s downstream. This distance-dependent bandwidth degradation is the fundamental economic driver behind Fiber-to-the-Home (FTTH) deployment.

3.1.4 Loading Coils (Pupin Coils)

In the early 20th century, **loading coils** (also called **Pupin coils** after Michael Pupin, 1900) were inserted in series along long subscriber loops to extend the voice-band transmission range. A loading coil is a discrete inductor (typically 88 mH inductance in the H-88 standard) inserted every 1830 m along the line.

Operating principle: The added series inductance resonates with the distributed shunt capacitance of the line, producing a **low-pass Butterworth filter response** that significantly flattens the attenuation across the voice band (300–3400 Hz), extending usable loop length from ≈5 km to ≈12 km. However, the loading coil introduces a **sharp cutoff above ≈3.4 kHz**, rendering the loop unusable for any broadband service. When DSL was introduced, loading

coils had to be physically located and removed (”de-loaded”) from all subscriber loops intended for DSL service — a major and costly network engineering task.

3.1.5 UTP, FTP, and STP Variants

- **UTP (Unshielded Twisted Pair):** No metallic shielding of any kind. The standard form for PSTN subscriber local loops and LAN structured cabling (Cat5e, Cat6, Cat6A). Cost-effective and flexible, but relies entirely on the twisting for EMI rejection. Limited performance in high-EMI environments.
- **FTP / ScTP (Foiled Twisted Pair / Screened):** A single metallic foil (aluminium, bonded to a drain wire) wraps the entire cable bundle. Provides an overall electrostatic shield against external interference without shielding individual pairs. Reduces alien crosstalk from external sources. Used in structured cabling for medium-EMI environments.
- **STP (Shielded Twisted Pair):** Individual metallic shields around each pair, plus an outer shield. Maximizes isolation between pairs (minimizes NEXT/FEXT) and provides best external EMI rejection. Used in demanding environments (industrial plants, alongside power cables). Heavier, stiffer, and more expensive to terminate correctly (requires proper grounding at both ends for effectiveness).

3.2 Coaxial Cable

3.2.1 Physical Structure

A **coaxial cable** (abbreviated “coax”) consists of concentric cylindrical conductors sharing a common geometric axis, hence the name. From center outward, the construction is:

1. **Inner conductor:** A solid or stranded copper (or copper-clad aluminum) wire at the center. Carries the signal.
2. **Dielectric:** A cylindrical layer of insulating material surrounding the inner conductor. Common dielectrics: solid polyethylene (PE), foamed PE (for lower loss), or polytetrafluoroethylene (PTFE, for high-temperature applications). The dielectric determines the characteristic impedance and significantly affects attenuation.

3. **Outer conductor (shield):** A cylindrical metallic layer — braided copper wire, bonded aluminum foil, or corrugated solid metal tube — surrounding the dielectric. Serves simultaneously as the return current path and as a complete electromagnetic shield.
4. **Outer jacket:** A protective plastic sheath (PVC, PE, or halogen-free low-smoke compound) protecting the cable from mechanical and environmental damage.

3.2.2 Electrical Characteristics and Advantages

- **Complete electromagnetic self-shielding:** The outer conductor fully encloses the electromagnetic field of the inner conductor: **no signal field exists outside the cable**, and no external EMI field can penetrate to the inner conductor. Coaxial cable neither radiates nor receives interference. This is a fundamental geometric property, independent of shielding quality.
- **High and controllable characteristic impedance:** $Z_0 = (138 / \sqrt{\varepsilon_r}) \times \log_{10}(D/d)$, where D = inner diameter of outer conductor, d = diameter of inner conductor, ε_r = relative permittivity of dielectric. Standard values: **50 Ω** (RF engineering, telecommunications, test equipment) and **75 Ω** (television distribution, cable TV, video). The precise geometry ensures a well-defined, frequency-stable impedance.
- **Wide bandwidth:** High-quality coaxial cable supports signal transmission from DC to several GHz. Standard CATV coax (RG-6): usable to ≈1 GHz. Precision test cables: beyond 18 GHz. This wide bandwidth made coaxial cable the dominant long-haul backbone medium for analog FDM telephony (see Section 3.2.3).
- **Lower attenuation than twisted pair at equivalent frequency:** The coaxial geometry minimizes dielectric loss and conductor loss. Attenuation is proportional to $\sqrt{f}$ (skin effect in inner conductor and braid). Typical value for standard 75 Ω CATV coax: ≈5 dB/100 m at 1 GHz.

3.2.3 Applications in Telephony

- **Analog FDM long-distance backbone:** From the 1940s through the late 1970s, large-diameter coaxial cables — carrying the ITU-T standardized FDM hierarchy (Basic Group: 12 ch; Super Group: 60 ch; Master Group: 300 ch; Super Master Group: 900 ch per coaxial pair) — formed the intercity and intercontinental backbone of analog

telephone networks. The AT&T L5 coaxial system (1974) carried 10,800 two-way telephone conversations on a single coaxial tube pair. Completely superseded by optical fiber from the late 1980s onward.

- **Hybrid Fiber-Coax (HFC) broadband access:** Cable television operators deploy coaxial cable for the final distribution segment from neighborhood optical nodes to subscriber premises, combined with optical fiber backhaul (hence "hybrid"). The DOCSIS 3.1/4.0 standard enables downstream speeds of 1–10 Gbit/s and upstream of 1–6 Gbit/s on the coaxial segment. VoIP telephone service is delivered over the same HFC infrastructure.
- **Intra-facility and RF connections:** Within telephone exchange buildings, between equipment racks, and for connecting RF and microwave components (antennas, filters, diplexers), coaxial cable remains the standard interconnect medium.

4. Wireless (Radio) Transmission

Wireless transmission employs **free-space electromagnetic wave propagation** as the transmission medium: no physical conductor connects the transmitter and receiver. This characteristic makes wireless systems uniquely valuable wherever physical cable installation is impractical, impossible, or uneconomical — across difficult terrain, over bodies of water, to mobile subscribers, and for emergency or temporary communications.

4.1 Fundamentals of Radio Wave Propagation

Radio waves are electromagnetic waves characterized by frequency f (Hz) and wavelength $\lambda = c/f$. Propagation behavior depends strongly on frequency and determines the applicability of each band to telephony:

Band	Frequency	Propagation Mechanism	Telephony Application
HF	3–30 MHz	Sky wave: ionospheric reflection; global range, variable quality	HF radiotelephony: maritime, aeronautical, remote-area voice (unreliable)
VHF	30–300 MHz	Line-of-sight (LoS) + diffraction; limited troposcatter	Land mobile radio (PMR); 1G analog cellular (NMT, AMPS); paging
UHF	300 MHz–3 GHz	Line-of-sight; significant diffraction around obstacles; multipath	2G/3G/4G/5G cellular; DECT cordless; Wi-Fi calling; IoT telephony
SHF (Microwave)	3–30 GHz	Strict line-of-sight; rain fade above 10 GHz; high directivity	Point-to-point microwave trunks; base station backhaul; satellite uplink/downlink
EHF (mm-Wave)	30–300 GHz	Very strict LoS; high O_2 and H_2O absorption; very high path loss	5G NR (FR2): small cells; Ka-band satellite; ultra-short-range backhaul

4.2 Point-to-Point Microwave Links

Point-to-point microwave radio links are the classic wireless medium for **inter-exchange trunk connections** and **base station backhaul**. They operate in the SHF band (4–40 GHz typical, licensed bands) between tower-mounted **directional parabolic dish antennas** that concentrate the radiated beam precisely toward the receiving station, minimizing interference and maximizing received power.

4.2.1 Link Budget and System Design

The fundamental design tool is the **link budget**: an accounting of all signal gains and losses from transmitter to receiver to ensure the received power exceeds the receiver's sensitivity threshold by a sufficient **system margin** under worst-case conditions.

- **Free-space path loss:** FSPL (dB) = $20\times\log_{10}(d) + 20\times\log_{10}(f) + 20\times\log_{10}(4\pi/c)$. Increases as the square of distance and the square of frequency. A 10 GHz link at 50 km experiences approximately 146 dB of free-space loss.
- **Antenna gain:** Large parabolic dish antennas provide **30–50 dBi of gain** in the main beam, partially compensating free-space loss. Gain √ aperture area × frequency.
- **Fresnel zone clearance:** For reliable LoS propagation, the **first Fresnel zone** (an ellipsoidal volume around the direct path) must be at least 60% clear of obstacles. This determines the minimum tower heights for a given hop length and terrain profile.
- **Fade margin:** The system is designed with a **fade margin of 20–40 dB** above the minimum receiver sensitivity to accommodate: (a) **rain attenuation** (significant above 10 GHz; up to 10–20 dB/km during heavy rain at 30 GHz); (b) **atmospheric multipath fading** (ducting, k-factor variation); (c) equipment aging. This margin ensures link availability exceeding 99.999% of the time (ITU-T G.826).

4.2.2 Modulation and Capacity

Modern **digital microwave** systems use **adaptive modulation**: under favorable propagation conditions, high-order modulation (256-QAM, 1024-QAM, 4096-QAM) is used to maximize spectral efficiency. Under degraded conditions (rain, fading), the system automatically falls back to more robust modulation (QPSK, 16-QAM) to maintain link availability at reduced capacity. Typical capacities range from **2 Mbit/s (one E1) to several Gbit/s** per hop, in licensed channel bandwidths of 3.5–112 MHz. **Dual-polarization**

(simultaneous transmission at horizontal and vertical polarizations on the same frequency) doubles the capacity of a single hop.

4.3 Satellite Telephony

A communication satellite acts as a **radio relay station in orbit**: it receives the uplink signal from a ground station, performs frequency conversion (and amplification in the satellite's transponder), and retransmits the downlink to one or more receiving ground stations. This architecture enables communication over **continental to global distances** without any ground infrastructure between endpoints.

4.3.1 Geostationary Orbit (GEO) Satellites

A GEO satellite orbits at exactly **35,786 km** above the equator, where orbital period = 24 hours. It appears stationary from Earth, allowing fixed directional antennas and simplifying network design. Three GEO satellites at 120° longitude spacing provide near-global coverage (excluding polar regions).

Critical limitation — latency: The round-trip propagation path is ≈72,000 km. At the speed of light: one-way delay ≈ 240 ms; round-trip delay (RTD) ≈ 480–600 ms including equipment delays. Consequences for telephony:

- **Echo:** The talker hears a delayed copy of their own voice (talker echo). ITU-T G.168 echo cancellers are **mandatory** on all GEO satellite voice circuits. The standard requires echo loss ≥37 dB for delays up to 160 ms.
- **Conversational awkwardness:** Half-second gap between utterance and response causes speakers to interrupt each other or pause unnaturally. Acceptable for occasional use; frustrating for sustained dialogue.
- **TCP/IP degradation:** TCP slow-start and congestion control are severely impacted by the large RTD. TCP performance enhancement proxies (PEP) are required for broadband data via GEO satellite.

GEO satellite telephony applications: **rural and remote telephony** in regions without terrestrial infrastructure (Africa, Arctic, Amazon basin, Pacific islands), **maritime and aeronautical communication**, television and radio program distribution, **VSAT** (Very Small Aperture Terminal) enterprise networks, and disaster recovery communications.

4.3.2 Low Earth Orbit (LEO) and Medium Earth Orbit (MEO)

LEO constellations operate at 160–2,000 km altitude. At 550 km (e.g., Starlink), one-way delay ≈ 2–3 ms — comparable to a long terrestrial fiber link. RTD ≈20–60 ms. This makes LEO satellite telephony and broadband entirely acceptable for real-time voice and video conferencing.

Technical requirements of LEO constellations:

- **Large constellation size:** A LEO satellite at 550 km is above the horizon of any ground point for only ≈5–10 minutes per pass. Continuous coverage requires hundreds to thousands of satellites (SpaceX Starlink: 6,000+ satellites; Iridium: 66 satellites at 780 km with inter-satellite links).
- **Inter-satellite links (ISL):** Iridium uses Ka-band ISLs to route calls directly between satellites, enabling truly global coverage without intermediate ground stations. Starlink Gen2 also employs ISLs.
- **Frequent handoffs:** Subscriber terminals must seamlessly transition between satellites every few minutes. Requires sophisticated mobility management, similar in concept to cellular handover.

MEO constellations (e.g., O3b/SES at 8,062 km) offer a compromise: RTD ≈70–150 ms (acceptable for most voice protocols without special echo cancellation) with fewer satellites than LEO (≈20 satellites for non-polar coverage).

4.4 Cellular Radio Telephony

Cellular radio is the technology underlying all **mobile telephone networks**. The foundational **cellular principle** was conceived by D.H. Ring and W.R. Young at Bell Labs in 1947 and first implemented commercially in 1979 (NTT, Japan) and 1983 (AMPS, USA). It is the most widely deployed telephone access technology in the world, with over 8.5 billion mobile subscriptions globally (2024).

4.4.1 The Cellular Principle

The cellular architecture solves the fundamental problem of **spectrum scarcity**: the available radio frequency spectrum is finite, yet the number of simultaneous telephone calls that must be supported is very large. The solution is **geographic frequency reuse**:

- The coverage area is divided into a mosaic of small geographic **cells**, each served by a **Base Transceiver Station (BTS)** with a low-power, directional or omnidirectional antenna.
- The available frequency spectrum is divided into a set of **channel groups**. Each cell is assigned one channel group.
- Cells that are separated by a sufficient **reuse distance** (determined by the frequency reuse factor N, the ratio of reuse distance to cell radius: $D/R = \sqrt{(3N)}$) can use **the same channel group simultaneously** without mutual co-channel interference exceeding acceptable limits.
- The total network capacity = (capacity per cell) × (number of cells). By making cells smaller (**cell splitting**), capacity can be increased without additional spectrum.

Cellular Analogy

Consider a city with 7 available radio channels. Scenario A (single transmitter): one powerful station covers the city using all 7 channels — maximum 7 simultaneous calls. Scenario B (cellular, N=7 reuse): divide the city into 70 hexagonal cells; assign each cell 1 channel; reuse each channel in every 7th cell. Result: up to 70 simultaneous calls (one per cell), a 10× capacity increase with the same spectrum allocation. Further cell splitting multiplies capacity proportionally.

4.4.2 Key Technical Elements

- **Multiple Access Schemes:** How simultaneous subscribers share the available spectrum within one cell: **FDMA** (1G: each call on a unique frequency), **TDMA** (2G/GSM: 8 time slots per 200 kHz carrier; DECT: 24 slots), **CDMA** (3G/UMTS: all calls share 5 MHz bandwidth, separated by orthogonal spreading codes; processing gain), **OFDMA** (4G LTE / 5G NR: flexible multi-carrier allocation of resource blocks per user, supports MIMO).
- **Handover (Handoff):** As a subscriber moves between cells, the active call transfers from one BTS to the next. **Hard handover** (break before make, GSM): brief interruption; **Soft handover** (make before break, WCDMA): simultaneous connection to multiple BTSs using macro diversity, improving SNR at cell edge.

- **Location management:** The network tracks the approximate location of idle mobile terminals through **Location Area** registration. When an incoming call arrives, the network pages all BTSs in the subscriber's last known Location Area to alert the terminal.
- **Evolution of generations: 1G** (analog FM: AMPS, NMT, TACS, 1979–1991) → **2G** (digital: GSM/GPRS, cdmaOne, 1991–2001) → **3G** (WCDMA/UMTS, CDMA2000, 2001–2010) → **4G** (LTE/LTE-A, all-IP, 2009–present) → **5G** (NR, sub-6 GHz and mmWave, 2019–present). Each generation increased spectral efficiency, data rates, and introduced new service categories.

5. Optical Fiber Transmission

Optical fiber is the dominant transmission medium for **long-haul, metropolitan, and increasingly access-network** telephony and broadband infrastructure. It constitutes a paradigm shift from electrical to photonic signal transmission, offering capabilities qualitatively superior to any electrical or radio medium in terms of bandwidth, attenuation per unit length, and electromagnetic immunity. Since the early 1990s, optical fiber has progressively replaced coaxial cable on backbone routes and copper on access routes, and this migration continues with the global rollout of Fiber-to-the-Home (FTTH).

5.1 Physical Principles: Total Internal Reflection

An optical fiber is a cylindrical waveguide made of **ultra-pure silica glass (SiO_2)** with a refractive index profile designed to confine and guide light along its axis. The structure consists of two concentric regions:

- **Core:** The central glass region, refractive index n_1. Light signals propagate here. Typical diameter: 9 µm (single-mode) or 50/62.5 µm (multimode).
- **Cladding:** A surrounding concentric glass layer, refractive index $n_2 < n_1$ (difference $\Delta n \approx 0.003–0.02$). Outer diameter: 125 µm (standard). Does not carry the optical signal but is essential for guiding it.

Light guidance is achieved by **Total Internal Reflection (TIR)**. When a light ray in the denser core (n_1) reaches the core-cladding interface at an angle of incidence **θ** (measured from the normal to the interface) greater than the **critical angle θc**:

> **$\theta_c = \arcsin(n_2 / n_1)$**
>
> ...the ray is completely reflected back into the core with zero energy loss to the cladding. By designing the fiber so that all injected rays exceed θ_c, light is guided along the fiber axis with very low loss through multiple reflections, even around gentle curves. The **Numerical Aperture** $NA = \sqrt{(n_1^2 - n_2^2)} \approx n_1 \times \sqrt{(2\Delta n)}$ defines the cone of input angles within which light is accepted and guided.

5.2 Single-Mode vs. Multimode Fiber

The concept of **guided modes** arises from the wave-optic (electromagnetic) analysis of the fiber. A mode is a stable spatial distribution of the optical field that propagates along the fiber without changing shape. The number of guided modes is determined by the **V-number** (normalized frequency):

> **$V = (\pi \times d \times NA) / \lambda$**
>
> Where d = core diameter, NA = Numerical Aperture, λ = operating wavelength. Single-mode condition: V < 2.405 (only the fundamental HE_{11} mode propagates). Multimode: V >> 2.405 (hundreds to thousands of modes).

Property	Single-Mode Fiber (SMF)	Multimode Fiber (MMF)
Core diameter	~9 µm	50 µm (OM3/4/5) or 62.5 µm (OM1/2)
Cladding diameter	125 µm (standard)	125 µm (standard)
Guided modes	1 (HE_{11} fundamental)	Hundreds to thousands (V-dependent)
Bandwidth × distance	Essentially unlimited (tens of THz·km)	Limited: ~200 MHz·km (OM1) to ~2 GHz·km (OM5)
Attenuation	~0.2 dB/km @ 1550 nm (minimum); ~0.35 dB/km @ 1310 nm	~2–3 dB/km @ 850 nm; ~0.8 dB/km @ 1300 nm
Max unrepeated distance	80–120 km (EDFA spacing); transoceanic with EDFA chains	~300 m @ 10 Gbit/s (OM3); ~2 km @ 1 Gbit/s (OM4)
Light source	Narrow-linewidth DFB laser (expensive; precise alignment)	LED or VCSEL (low cost; relaxed alignment tolerance)
Primary ITU-T standard	G.652 (standard SMF); G.654 (ultra-low loss); G.657 (bend-insensitive)	G.651.1 (50 µm OM2–OM5)

Property	Single-Mode Fiber (SMF)	Multimode Fiber (MMF)
Primary application	Long-haul backbone; metro networks; FTTH; submarine cables	LAN, data center intra-facility; structured cabling

5.3 Attenuation in Optical Fiber

The remarkably low attenuation of modern silica fiber (≈0.2 dB/km) is the result of decades of materials science advancement. Attenuation arises from three physical mechanisms:

- **Rayleigh scattering:** Light is scattered by microscopic refractive index fluctuations (density variations frozen into the glass during manufacture). Rayleigh scattering attenuation $\propto 1/\lambda^4$: it dominates at shorter wavelengths and decreases rapidly as wavelength increases. Sets the minimum loss limit for conventional silica fiber.
- **Infrared absorption:** Silica glass molecules absorb photon energy corresponding to their vibrational modes at wavelengths above ~1700 nm. This sets the long-wavelength limit of usable transmission and causes a rapid rise in attenuation above 1600 nm.
- **OH^- (hydroxyl) ion absorption:** Trace water contamination in the glass creates absorption peaks, most significantly at 1383 nm (the **water peak**). Modern **low-water-peak (LWP) fiber** (ITU-T G.652.D, G.654) suppresses this peak, opening the full E-band (1360–1460 nm) for WDM use.

The combination of these mechanisms produces **low-loss transmission windows**:

- **O-band (Original):** 1260–1360 nm. Attenuation ≈0.35 dB/km. Zero chromatic dispersion of standard SMF at ≈1310 nm. Used for metro and access DWDM.
- **C-band (Conventional):** 1530–1565 nm. Minimum attenuation ≈0.19 dB/km. Coincides with EDFA gain bandwidth. The **dominant band for all long-haul DWDM** transmission.
- **L-band (Long):** 1565–1625 nm. Attenuation ≈0.20–0.25 dB/km. Used for extended DWDM capacity (L-band EDFAs and Raman amplifiers).

5.4 Dispersion in Optical Fiber

Dispersion is the **temporal broadening of optical pulses** as they propagate along the fiber. Because a real optical pulse is not perfectly monochromatic, different spectral components travel at slightly different velocities, causing the pulse to spread in time. Pulse broadening limits the maximum achievable **bit rate × distance product** before inter-symbol interference (ISI) degrades the bit error ratio.

- **Modal dispersion (MMF only):** Different modes propagate at different group velocities (different ray path lengths). High-order modes travel longer paths and arrive later, broadening the pulse. Eliminated in SMF by the single-mode condition ($V < 2.405$). **Graded-index MMF** (parabolic refractive index profile) reduces modal dispersion by 100× compared to step-index MMF by equalizing mode group velocities.
- **Chromatic dispersion (SMF):** Arises from two sub-effects: (a) **Material dispersion**: the refractive index of silica depends on wavelength (described by the Sellmeier equation), causing different wavelength components to propagate at different speeds. (b) **Waveguide dispersion**: the effective index of the guided mode depends on wavelength through the guidance geometry (V-number). Material and waveguide dispersion partially cancel at the **zero-dispersion wavelength λ_ZD**. For standard G.652 SMF: $\lambda_ZD \approx 1310$ nm. Dispersion-shifted fiber (DSF, G.653) moves λ_ZD to 1550 nm for use with early single-channel systems; non-zero dispersion-shifted fiber (NZDSF, G.655) provides small non-zero dispersion at 1550 nm to suppress four-wave mixing in DWDM.
- **Polarization Mode Dispersion (PMD):** Real SMF fibers have slight geometric imperfections and asymmetric mechanical stress, causing the two orthogonal polarization states of the single guided mode to travel at slightly different velocities (differential group delay, DGD). PMD is statistical, temperature-dependent, and becomes a significant impairment at bit rates ≥10 Gbit/s over long distances. Managed by low-PMD fiber specifications (G.654.E) and electronic PMD compensation at the receiver.

5.5 Optical Amplification: The EDFA

The **Erbium-Doped Fiber Amplifier (EDFA)**, invented in 1987 by David Payne (Southampton) and Emmanuel Desurvire (Bell Labs), is the enabling technology of modern long-haul and submarine fiber communications. An EDFA consists of a short segment (≈1–10 m) of silica fiber doped with erbium ions (Er^{3+}), excited (pumped) by a high-power semiconductor laser at 980 nm or 1480 nm.

Operating principle: Pump photons excite erbium ions to a higher energy state. When signal photons in the C-band (1530–1565 nm) pass through the erbium-doped fiber, they trigger **stimulated emission**: excited erbium ions emit additional photons at the **same wavelength, phase, and polarization** as the signal, amplifying it. This is **all-optical amplification** — no optical-to-electrical conversion is required.

> ☑ **Why the EDFA is Transformative**
>
> *Before the EDFA (pre-1990), long-haul fiber links required optoelectronic repeaters every 40–80 km: the optical signal was detected, converted to an electrical signal, regenerated, and retransmitted. Each repeater handled only one wavelength channel. The EDFA simultaneously amplifies ALL wavelength channels within its C-band gain bandwidth (≈35 nm), enabling Dense WDM (DWDM) systems to carry 80–160 or more independent wavelength channels on a single fiber pair. Combined with Raman amplification and advanced modulation (DP-QPSK, DP-16QAM, DP-64QAM), current commercial submarine systems achieve per-fiber capacities exceeding 50 Tbit/s.*

5.6 Dense Wavelength Division Multiplexing (DWDM)

DWDM extends the FDM concept — studied in Chapter 1 for analog telephony — into the optical domain. Multiple independent optical carriers (wavelength channels, also called **lambdas**) are simultaneously transmitted on a single fiber pair, each at a distinct wavelength within the C-band (or L-band).

- **Channel spacing:** ITU-T G.694.1 standardizes DWDM channel grids: 100 GHz spacing (standard, 80 channels in C-band), 50 GHz spacing (160 channels), and flexible grid (12.5 GHz granularity for spectrum-efficient networking).
- **Per-channel modulation:** Each wavelength channel carries a modulated optical signal. Modern coherent systems use **Dual-Polarization QPSK (DP-QPSK)** at 100 Gbit/s or **DP-16QAM** at 200 Gbit/s per channel. Advanced probabilistic constellation shaping achieves near-Shannon-limit spectral efficiency.
- **Total capacity:** A modern C+L band DWDM system with 96 channels at 400 Gbit/s achieves approximately **38.4 Tbit/s per fiber pair**. With spatial division multiplexing (multi-core or multi-mode fiber), aggregate capacities exceeding 1 Pbit/s per fiber have been demonstrated in research.

5.7 FTTH and Passive Optical Networks (PON)

Fiber-to-the-Home (FTTH) eliminates the copper local loop, replacing it with an optical fiber connection all the way to the subscriber premises. The dominant FTTH architecture is the **Passive Optical Network (PON)**, standardized by ITU-T (G-PON series) and IEEE (802.3ah EPON series):

- **Optical Line Terminal (OLT):** Located at the telephone exchange (central office). Terminates the optical network and connects to the IP/telephony core network.
- **Passive optical splitter:** A wavelength-independent optical power splitter (1:16, 1:32, or 1:64) installed in an outdoor optical distribution cabinet or joint, at a distance of 10–20 km from the OLT. Requires no power supply (“passive”). Splits the downstream signal to all subscribers and combines the upstream.
- **Optical Network Unit/Terminal (ONU/ONT):** Installed at the subscriber premises. Converts the optical signal to electrical for in-home distribution (Ethernet, telephony FXS port, CATV RF).
- **Capacity:** GPON (G.984): 2.5 Gbit/s downstream / 1.25 Gbit/s upstream shared per splitter. XGS-PON (G.9807): 10 Gbit/s symmetric. NG-PON2 (G.989): 40 Gbit/s downstream (4 wavelengths × 10 Gbit/s). 50G-PON (G.9804): 50 Gbit/s symmetric, currently deploying.

6. Comparative Analysis of Transmission Media

The table below applies all evaluation criteria of Section 2 to the four principal media, providing a concise engineering reference for media selection decisions.

Criterion	Twisted Pair (UTP)	Coaxial Cable	Wireless / Radio	Optical Fiber (SMF)
Bandwidth	Narrow: 3.1 kHz (POTS) up to ~35 MHz (VDSL2)	Wide: DC to several GHz (coax geometry)	Spectrum-limited; per-channel BW from 200 kHz (GSM) to 400 MHz (5G NR)	Enormous: >10 THz/fiber; 80–160+ DWDM channels
Attenuation	High: ~20 dB/km @ 1 MHz. Strongly distance-limited.	Medium: ~5 dB/100 m @ 1 GHz. Better than UTP at high freq.	Free-space path loss $\sqrt{d^2 \times f^2}$. Rain fade above 10 GHz.	Extremely low: ~0.2 dB/km @ 1550 nm. 80–120 km EDFA spacing.
Max Distance (unrepeated)	~5 km (POTS); ~0.3–5 km (xDSL, freq-dependent)	~1–10 km (distribution); ~50 km (backbone, historical)	50–80 km/hop (microwave); global (satellite)	80–120 km (EDFA spacing); transoceanic chains
EMI Susceptibility	High. Twisting reduces; shielding (FTP/STP) helps.	Very low. Self-shielded by outer conductor geometry.	Inherent; operates in shared RF environment; multipath.	Immune. No electromagnetic coupling. Ideal near HV lines, railways.

Criterion	Twisted Pair (UTP)	Coaxial Cable	Wireless / Radio	Optical Fiber (SMF)
Security	Low: passive or galvanic tapping straightforward.	Medium: outer conductor must be penetrated.	Low: open-space broadcast. Requires crypto encryption.	High: tap causes measurable loss; OTDR-detectable.
Cost profile	Low CapEx (copper + simple tools). High OpEx at high freq. (crosstalk mgmt).	Medium CapEx. Low OpEx (passive medium).	Medium–High CapEx (towers, BTSs). High OpEx (spectrum fees, power).	Medium–High CapEx (civil works). Very low OpEx. Highly scalable.
Primary network use	Subscriber local loop (POTS, xDSL, FTTC/VDSL)	HFC broadband (CATV+VoIP); historical backbone	Mobile access (all generations); rural backhaul; maritime/satellite	Long-haul backbone; metro; FTTH; submarine cables

7. Chapter Summary: Key Concepts to Remember

Concept / Term	Essential Definition / Key Point
Shannon's Capacity Theorem	$C = B \times \log_2(1 + S/N)$. Fundamental upper bound on bit rate; bandwidth and SNR are the only two available levers.
Attenuation (dB/km)	Signal power loss per unit length. Increases with frequency in copper. Determines maximum repeater spacing and thus infrastructure cost.

Concept / Term	Essential Definition / Key Point
Twisted Pair — Common-Mode Rejection	Differential receiver rejects equally-induced noise. Twisting equalizes exposure; shorter lay = better rejection. Principle of balanced transmission.
Loading Coils (Pupin)	Series inductors inserted every 1.83 km. Extend voice loop to ~12 km but introduce sharp cutoff at 3.4 kHz. Must be removed for DSL.
Coaxial Cable — Self-Shielding	Outer conductor encloses EM field completely: no radiation, no external EMI coupling. $Z_0 = 50\ \Omega$ (telecom) or 75 Ω (CATV). High bandwidth (DC to GHz).
Skin Effect	At high frequencies, AC current concentrates near the conductor surface, reducing effective cross-section and increasing resistance. Causes f-dependent attenuation increase in copper.
Total Internal Reflection (TIR)	Complete reflection of light at core-cladding interface when angle of incidence > $\theta_c = \arcsin(n_2/n_1)$. Physical basis of optical fiber light guidance.
Numerical Aperture (NA)	$NA = \sqrt{(n_1^2 - n_2^2)}$: characterizes the acceptance cone angle and the number of guided modes (V-number).
Single-Mode Fiber (SMF)	Core ~9 µm; $V < 2.405$; one guided mode; eliminates modal dispersion; attenuation ~0.2 dB/km @ 1550 nm; standard for long-haul and FTTH.
Multimode Fiber (MMF)	Core 50–62.5 µm; many modes; limited by modal dispersion. Used in LAN, data center intra-building links up to ~2 km.
Chromatic Dispersion	Pulse broadening due to wavelength-dependent group velocity in SMF. $\lambda_ZD \approx 1310$ nm (G.652). Managed by DCF modules or electronic dispersion compensation (EDC).

Concept / Term	Essential Definition / Key Point
EDFA	Erbium-Doped Fiber Amplifier: all-optical C-band amplification via stimulated emission. Amplifies ALL DWDM channels simultaneously. Spacing: 80–120 km. Enables transoceanic links.
DWDM	Dense WDM: 80–160+ wavelengths per fiber, ITU-T 50/100 GHz spacing. Each lambda carries 100–400 Gbit/s. Total fiber capacity: tens of Tbit/s.
GEO Satellite Latency	Altitude 35,786 km → one-way delay ~240 ms; RTD ~480–600 ms. Requires mandatory ITU-T G.168 echo cancellers. Acceptable for broadcast, VSAT, maritime; problematic for voice.
Cellular Principle	Geographic frequency reuse: identical frequencies used in non-adjacent cells separated by reuse distance $D = R\sqrt{(3N)}$. Enables capacity scaling without additional spectrum.
Passive Optical Network (PON)	FTTH architecture: OLT → passive 1:32/1:64 splitter (no field power required) → ONU at subscriber. GPON: 2.5/1.25 Gbit/s; XGS-PON: 10 Gbit/s symmetric; 50G-PON: deploying.

Looking Ahead: Chapter 3

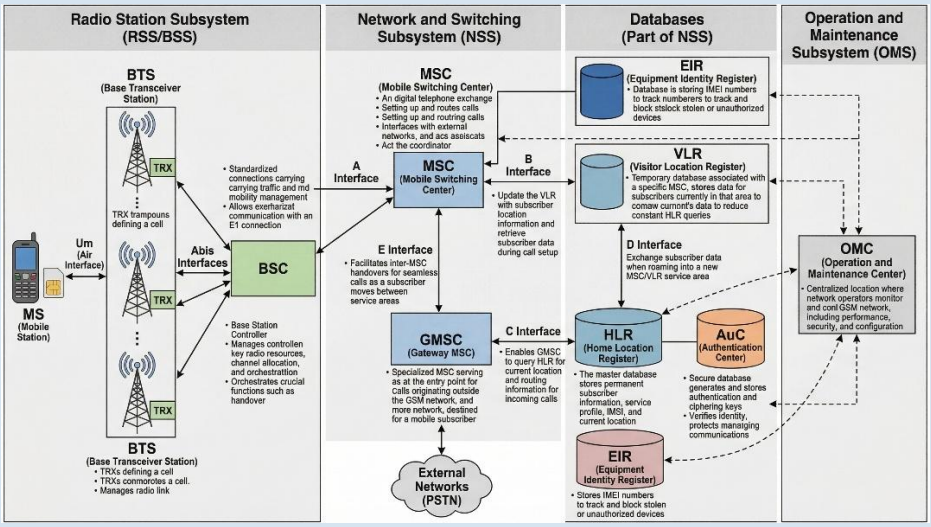
Chapter 3 will introduce Digital Telephony and Pulse Code Modulation (PCM). Building directly on the transmission media studied in this chapter, we will examine how the Nyquist sampling theorem enables analog voice signals to be converted into digital binary streams, how PCM regenerative repeaters overcome the noise accumulation problem of analog copper and wireless links, and how Time Division Multiplexing (TDM) and the E1/T1 plesiochronous digital hierarchy (PDH) organize these digital bit streams for efficient transport over the electrical conductors and optical fibers covered here.

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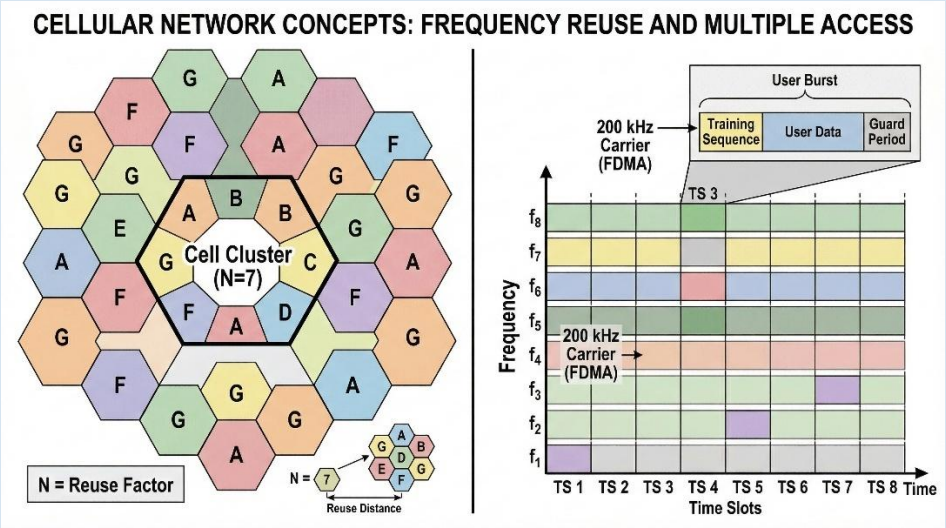
CHAPTER 3

Digital Cellular GSM Telephony:

Networks, Protocols, Architecture, Equipment, Diagrams, and Measurement



(1) The full GSM network architecture diagram showing MS → BTS → BSC → MSC → PSTN/HLR/VLR/AuC/EIR with labelled interfaces (Um, Abis, A, B, C, D, E).



(2) A hexagonal cell cluster diagram illustrating frequency reuse (N=7). (3) A TDMA/FDMA grid showing 8 time slots per 200 kHz carrier.

1. Introduction to GSM

1.1 Pedagogical Objectives

Upon successful completion of this chapter, the student will be able to:

- **Describe** the overall GSM network architecture and identify its three principal subsystems and their constituent network elements.
- **Explain** the function of each GSM network entity (MS, BTS, BSC, MSC, HLR, VLR, AuC, EIR, OMC) and the interfaces that connect them.
- **Explain** the FDMA/TDMA multiple-access scheme of GSM and the frequency reuse principle as applied to cellular network planning.
- **Trace** the signaling flow for a Mobile-Originated Call (MOC) and a Mobile-Terminated Call (MTC) step by step through the network.
- **Describe** the GSM three-layer protocol stack (Physical, Data Link, Network), the role of each layer on the Um and Abis interfaces, and the role of SS7 signaling in the NSS.
- **Identify and interpret** the key GSM radio performance indicators: RSSI, BER, RxQual, CDR, and CSSR, and explain their engineering significance.

1.2 The GSM Standard: Context and Significance

The **Global System for Mobile Communications (GSM)** is the world's first fully standardized, open, digital cellular telephone system. It constitutes the defining technology of the **Second Generation (2G)** of mobile telecommunications, succeeding the incompatible proprietary analog first-generation systems (AMPS in North America, NMT in Scandinavia, TACS in the United Kingdom) which had proliferated in the 1980s.

GSM was developed by the **Groupe Spécial Mobile** committee of CEPT (European Conference of Postal and Telecommunications Administrations), established in 1982. The first GSM call was made in 1991. The standard was subsequently adopted worldwide and transferred to the **3rd Generation Partnership Project (3GPP)** in 1998 for ongoing maintenance and evolution. At its peak (early 2010s), GSM served over **5 billion subscribers** across more than 210 countries, making it the most widely deployed telecommunications standard in history.

GSM introduced several fundamental advances over analog 1G systems:

- **Digital voice encoding:** Analog speech is converted to digital data using a **speech codec** (Regular Pulse Excited – Long Term Prediction, RPE-LTP) at 13 kbit/s (Full-Rate), enabling error correction, encryption, and more efficient spectrum use.
- **TDMA/FDMA multiple access:** Multiple users share each radio carrier frequency by occupying distinct time slots, multiplying spectral efficiency compared to analog FDMA systems.
- **International roaming:** Open, standardized interfaces between network elements from different manufacturers enable a subscriber's handset to operate in any GSM network worldwide — a capability absent in 1G.
- **Security:** Authentication of the subscriber (preventing SIM card cloning), encryption of the radio interface (preventing eavesdropping on the air interface), and IMEI-based equipment validation.
- **SMS (Short Message Service):** A data channel for short text messages, originally a secondary feature that became the dominant non-voice application of mobile telephony until the smartphone era.

GSM Frequency Bands (Principal)

GSM-900: Uplink 890–915 MHz / Downlink 935–960 MHz. 124 carrier pairs at 200 kHz spacing. Original European band. Better propagation, larger cell radius.

DCS-1800: Uplink 1710–1785 MHz / Downlink 1805–1880 MHz. 374 carrier pairs. Higher capacity; smaller cells (urban micro-cells).

PCS-1900: Uplink 1850–1910 MHz / Downlink 1930–1990 MHz. 299 carrier pairs. North American deployment (USA, Canada).

2. GSM Architecture and Equipment

The GSM network is organized into three major functional subsystems, each with distinct responsibilities and well-defined standardized interfaces. This modular architecture allows network elements from different vendors to interoperate, provided they conform to the relevant 3GPP technical specifications.

Subsystem	Principal Elements	Primary Function
Radio Station Subsystem (RSS/BSS)	MS, BTS, BSC	Radio access: manage the air interface between the mobile and the fixed network
Network and Switching Subsystem (NSS)	MSC, HLR, VLR, AuC, EIR	Call routing, subscriber mobility management, authentication, and PSTN interconnection
Operation and Maintenance Subsystem (OMS)	OMC-R, OMC-S	Centralized monitoring, fault management, performance data collection, software management

2.1 Radio Station Subsystem (RSS / BSS)

The **Base Station Subsystem (BSS)** — also called the Radio Station Subsystem (RSS) — is the portion of the GSM network directly responsible for the **radio access function**: establishing, maintaining, and releasing radio links between mobile terminals and the fixed network. It consists of three components: the Mobile Station, the Base Transceiver Station, and the Base Station Controller.

2.1.1 Mobile Station (MS)

The **Mobile Station (MS)** is the subscriber's terminal equipment. It consists of two logically and physically separable components:

- **Mobile Equipment (ME):** The handset hardware — the radio transceiver (covering the GSM band), baseband processor, display, keypad, microphone, and loudspeaker. Each ME is uniquely identified by its **International Mobile Equipment Identity (IMEI)**, a 15-digit number that can be checked against the EIR to detect stolen or non-compliant handsets.
- **Subscriber Identity Module (SIM):** A removable smart card (ISO 7816-compliant) containing the subscriber's identity and security credentials. Key stored data: **IMSI** (International Mobile Subscriber Identity — the permanent network identity, 15 digits), **Ki** (individual authentication key, 128-bit, stored securely and never transmitted), and the A3/A8 authentication and key generation algorithms. The SIM enables **operator-subscriber binding** independent of the handset, and is the basis for international roaming.

MS functional responsibilities: Speech encoding (RPE-LTP codec), channel coding and interleaving, GMSK modulation, power control (adjusting transmit power per BSC/BTS instructions), timing advance (pre-timing transmissions to compensate for propagation delay at different distances from BTS), monitoring neighboring cell signal strengths (for handover decisions), and executing authentication and ciphering procedures.

2.1.2 Base Transceiver Station (BTS)

The **Base Transceiver Station (BTS)** is the radio equipment installed at the base of an antenna tower (or rooftop site) that serves a single **cell** (or up to three 120° sectors for a sectorized cell). The BTS is the network-side endpoint of the **Um interface** (the air interface).

- **Transceiver units (TRX):** Each TRX handles one 200 kHz carrier frequency, providing 8 TDMA time slots. A BTS may have 1–6 or more TRXs, each on a different carrier, thus supporting 8–48+ simultaneous traffic channels per cell (before frequency hopping and interference coordination).
- **Radio functions:** GMSK modulation and demodulation of the radio signal, channel encoding and decoding (convolutional coding, interleaving, Viterbi decoding), power amplification of the downlink signal, and reception of the uplink signal with diversity combining.

- **Measurement reporting:** The BTS measures the received signal quality from each active MS (uplink RSSI and BER measurements) and reports these to the BSC to support handover and power control decisions.
- **Transcoding:** In some configurations, the BTS performs **transcoding** between the GSM RPE-LTP speech codec (13 kbit/s) and the 64 kbit/s PCM format used on the Abis and A interfaces (using a **TRAU** — Transcoder/Rate Adaptor Unit — which may be co-located with the BTS or remotely hosted at the BSC).

2.1.3 Base Station Controller (BSC)

The **Base Station Controller (BSC)** is the intelligent control element of the BSS, typically controlling between 10 and several hundred BTSs. It performs the **radio resource management** functions for all cells under its control:

- **Frequency and timeslot allocation:** Assigns specific carrier frequencies and TDMA time slots to active calls and signaling channels within its cell domain.
- **Handover decision and execution (intra-BSC):** Based on measurement reports from the MS and BTSs, the BSC decides when to initiate handover between two BTSs under its control and executes the radio resource reassignment without MSC involvement (seamless to the MSC).
- **Power control:** Commands both the BTS (downlink power) and the MS (uplink power) to adjust transmit power to the minimum level required for acceptable link quality, reducing co-channel interference and extending handset battery life.
- **Paging:** Distributes paging messages received from the MSC to the appropriate BTSs within the Location Area of the called subscriber.
- **TRAU management:** Controls the Transcoder/Rate Adaptor Units, which convert between GSM codec rates and standard 64 kbit/s PCM for transport to the MSC.

GSM Air Interface and BSS Interfaces

Um (Air Interface): Between MS and BTS. Uses FDMA/TDMA/GMSK on the licensed GSM frequency bands. Defined by 3GPP TS 45.xxx series. The most critical and bandwidth-limited interface in the network.

Abis Interface: Between BTS and BSC. Typically implemented over E1 (2.048 Mbit/s) PCM links or microwave backhaul. Carries traffic channels (TRAU frames at 16 kbit/s per channel), OML (Operations and Maintenance Link), and RSL (Radio Signaling Link). Defined in 3GPP TS 48.051–48.060.

A Interface: Between BSC and MSC. Implemented over E1 links using the SS7 BSSAP (Base Station System Application Part) protocol. Carries both user traffic (64 kbit/s PCM voice channels) and BSSAP signaling (BSS Management Application Part — BSSMAP, and Direct Transfer Application Part — DTAP).

2.2 Network and Switching Subsystem (NSS)

The **Network and Switching Subsystem (NSS)** is the core of the GSM network. It performs **call routing and switching** between mobile subscribers and between the GSM network and the Public Switched Telephone Network (PSTN) or other mobile networks. It also manages **subscriber mobility** (tracking where subscribers are located), **authentication**, and **security**. The NSS is entirely digital, using SS7 signaling protocols for inter-element communication.

2.2.1 Mobile Switching Center (MSC)

The **Mobile Switching Center (MSC)** is the central and most critical element of the NSS. It is a high-capacity digital telephone exchange (derived from the fixed PSTN exchange architecture) augmented with mobile-specific capabilities:

- **Call routing and switching:** Routes calls between mobile subscribers served by the MSC and between MSC and other networks. Uses the dialed number (MSISDN) to query the HLR and determine the current location of the called subscriber.

- **Mobility management:** Processes **location update** requests from MSs as they move between Location Areas. Coordinates with the VLR for registration of visiting subscribers.
- **Handover control (inter-MSC):** When an active call must be handed over between cells served by different MSCs, the **anchor MSC** (serving the call origin) coordinates with the **target MSC** over the E interface (SS7/MAP).
- **Interconnection:** The **Gateway MSC (GMSC)** is the MSC responsible for receiving incoming calls from external networks (PSTN, other operators) and routing them to the MSC currently serving the called subscriber, by querying the HLR.
- **Authentication and ciphering:** Initiates and coordinates the authentication procedure (using AuC-generated triplets) and the activation of ciphering on the Um interface.

2.2.2 Home Location Register (HLR)

The **Home Location Register (HLR)** is the **master database** for all subscribers whose **home network** is this operator. It is a highly available, distributed database. Key stored data:

- **IMSI** (permanent subscriber identity) and **MSISDN** (the subscriber's telephone number, used by calling parties).
- **Service profile:** List of activated supplementary services (call forwarding, call waiting, conference calling, etc.) and subscribed bearer services.
- **Current location pointer:** The address of the **VLR** currently serving the subscriber (updated whenever the subscriber registers in a new MSC/VLR area). This allows the GMSC to route incoming calls to the correct MSC.
- **Roaming status:** Whether the subscriber is currently in the home network or roaming in a visited network, and which visited VLR/MSC is serving them.

2.2.3 Visitor Location Register (VLR)

The **Visitor Location Register (VLR)** is a database co-located with (or integrated into) the MSC. It is a **temporary local cache** of subscriber data for all MSs currently located within the MSC's coverage area, whether home subscribers or roamers from other networks. When an MS registers in a new MSC/VLR area, the VLR retrieves a copy of the subscriber's service profile from the HLR (via the C interface, using SS7/MAP) and creates a local record.

Why the VLR? Having subscriber data locally in the VLR avoids querying the remote HLR for every call setup and location update, dramatically reducing HLR load and signaling delay. The VLR also assigns a **Temporary Mobile Subscriber Identity (TMSI)** to protect the subscriber's IMSI from interception on the radio interface: subsequent radio transactions use the TMSI instead of the IMSI.

2.2.4 Authentication Center (AuC)

The **Authentication Center (AuC)** is a highly secure functional entity, usually co-located with and tightly integrated into the HLR. It is responsible for generating the **security triplets** used to authenticate each subscriber and generate the session encryption key:

- **Ki**: The 128-bit individual authentication key, unique to each SIM, stored securely in both the SIM and the AuC.
- **RAND**: A 128-bit random number generated by the AuC.
- **SRES**: Signed RESponse — 32-bit value computed by both the AuC and the SIM using algorithm A3: SRES = A3(Ki, RAND). Used to verify that the SIM holds the correct Ki.
- **Kc**: 64-bit cipher key computed using algorithm A8: Kc = A8(Ki, RAND). Used to encrypt the Um interface with algorithm A5.

The AuC pre-computes batches of **(RAND, SRES, Kc) triplets** for each subscriber and delivers them to the VLR. The VLR uses one triplet per authentication event. The Ki never leaves the AuC/SIM pair, making the system resistant to interception-based cloning attacks.

2.2.5 Equipment Identity Register (EIR)

The **Equipment Identity Register (EIR)** is a database that stores lists of **IMEI numbers** classified into three categories:

- **White List:** IMEIs of type-approved, valid mobile equipment. Calls from these devices are allowed normally.
- **Grey List:** IMEIs of devices under surveillance (e.g., suspected stolen, software fault). Calls may be allowed but flagged for monitoring. Implementation is operator-discretionary.

- **Black List:** IMEIs of stolen, barred, or non-type-approved devices. Calls from blacklisted IMEIs are rejected. Shared between operators through the IMEI Database (IMEIDB) / GSMA central database.

2.2.6 NSS Interfaces Summary

Interface	Between	Protocol	Function
A	BSC ↔ MSC	SS7 / BSSAP (BSSMAP + DTAP)	Traffic channels + BSS-MSC signaling
B	MSC ↔ VLR	Internal (usually co-located)	MSC queries VLR for subscriber data
C	GMSC ↔ HLR	SS7 / MAP	Routing enquiry for incoming calls
D	HLR ↔ VLR	SS7 / MAP	Location update; service profile transfer
E	MSC ↔ MSC	SS7 / MAP	Inter-MSC handover coordination
F	MSC ↔ EIR	SS7 / MAP	IMEI check (equipment identity verification)
G	VLR ↔ VLR	SS7 / MAP	TMSI reallocation between VLRs
H	HLR ↔ AuC	Proprietary (tight coupling)	Security triplet (RAND/SRES/Kc) retrieval

2.3 Operation and Maintenance Subsystem (OMS)

The **Operation and Maintenance Subsystem (OMS)** provides centralized management and visibility of the entire GSM network. It consists of one or more **Operation and Maintenance Centers (OMC)**, typically divided into:

- **OMC-R (Radio):** Manages the BSS elements (BTSs and BSCs). Functions: remote software loading to BTS/BSC, fault alarm collection and display, performance statistic collection (traffic, handover success rates, BER distributions), cell parameter configuration (transmit power, neighbor cell lists, handover thresholds), and remote BTS restart.
- **OMC-S (Switching):** Manages the NSS elements (MSC, HLR, VLR, AuC, EIR). Functions: subscriber provisioning (activating/deactivating subscriptions, modifying service profiles), call data record (CDR) collection for billing, fault management of MSC and database elements, and software upgrade management.

The OMC communicates with network elements over a dedicated **Operations Support System (OSS)** network, using protocols conforming to the **TMN (Telecommunications Management Network)** framework (ITU-T M.3000 series), typically implemented via CORBA, SNMP, or proprietary Element Manager protocols over a TCP/IP management LAN.

3. Key GSM Concepts and Principle Diagrams

3.1 Frequency Reuse and Cell Planning

The GSM radio spectrum is a finite and expensive resource. The cellular network architecture maximizes spectral efficiency through **geographic frequency reuse**: the same set of radio frequencies is assigned to multiple non-adjacent cells simultaneously. This concept — introduced conceptually in Chapter 2 for cellular systems in general — is here formalized for GSM-specific planning.

3.1.1 The Reuse Cluster

The coverage area is divided into **hexagonal cells** (an idealized geometric model). The available carrier frequencies are grouped into **N frequency sets** forming a **cluster** of N cells. The cluster tiles the coverage area periodically: every cell in the same position within each cluster uses **the same frequency set**. The minimum separation between co-channel cells is:

> **$D/R = \sqrt{(3N)}$**
>
> Where D = minimum co-channel reuse distance, R = cell radius, N = cluster size (reuse factor). Common GSM values: N=4 (D/R=3.46), N=7 (D/R=4.58), N=12 (D/R=6.00). Larger N → greater separation → less co-channel interference → but fewer frequencies per cell → less capacity per cell. The trade-off between interference protection and per-cell capacity is the central challenge of GSM frequency planning.

Frequency Hopping (FH): GSM optionally employs **slow frequency hopping** (SFH): the MS and BTS change carrier frequency at each TDMA frame (217 hops/second) following a pseudo-random hopping sequence defined by parameters broadcast by the BTS. FH provides **frequency diversity** (averaging out the fading variations of any single carrier) and **interference diversity** (interference from co-channel cells is spread across many carriers rather than continuously degrading one channel). FH effectively reduces the required cluster size (enabling N=1 or N=3) in well-planned networks with interference averaging.

3.2 Multiple Access: FDMA + TDMA

GSM employs a two-dimensional multiple access scheme combining **Frequency Division Multiple Access (FDMA)** and **Time Division Multiple Access (TDMA)**. Understanding this scheme is essential for correctly interpreting GSM capacity calculations and channel assignments.

3.2.1 FDMA Layer

In GSM-900, the 25 MHz uplink spectrum (890–915 MHz) and the 25 MHz downlink spectrum (935–960 MHz) are each divided into **124 carrier channels** of **200 kHz bandwidth** each (plus 1 guard channel), numbered ARFCNs 1–124. Each cell in the network is assigned a subset of these ARFCNs (determined by the frequency plan). The uplink and downlink of each carrier pair are separated by **45 MHz duplex offset**, allowing simultaneous bidirectional transmission without interference between the transmit and receive chains.

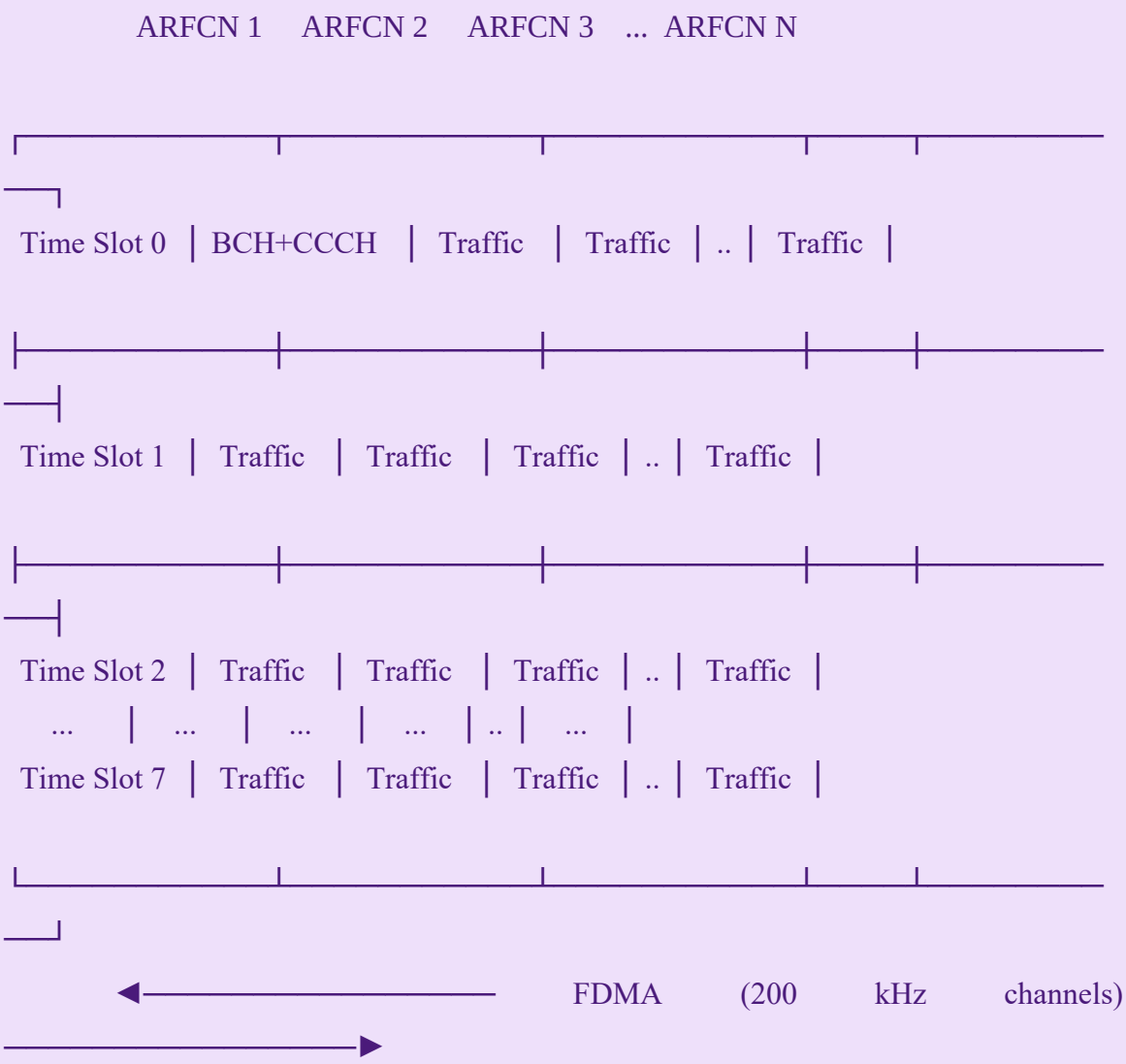
3.2.2 TDMA Layer

Each 200 kHz carrier frequency is further shared in time by **8 users** via TDMA. The carrier is divided into frames of duration **4.615 ms**. Each frame contains **8 time slots** (TSs) of 576.9 µs each (approximately 156.25 bits per slot at 270.833 kbit/s gross bit rate). Each active connection is assigned one specific time slot in each frame for its exclusive use.

A **Physical Channel** in GSM is therefore uniquely defined by two parameters: its **ARFCN** (carrier frequency) and its **time slot number** (0–7). This yields 8 physical channels per carrier, and up to 8 × 124 = 992 physical channels in the full GSM-900 band.

Principle Diagram: FDMA + TDMA Grid (GSM-900 Example)

Each column = one 200 kHz carrier (ARFCN). Each row = one TDMA time slot (0–7). Each cell = one Physical Channel.



ARFCN 1 Time Slot 0: assigned to BCCH + CCCH (broadcast + common control channels) on the **BCCH carrier** — always transmitted at constant power, never used for traffic. All other physical channels: available for Traffic Channels (TCH) or dedicated control channels (SDCCH, SACCH, FACCH).

3.2.3 GSM Logical Channels

GSM maps **logical channels** onto physical channels. The two principal categories are:

- **Traffic Channels (TCH):** Carry encoded speech (TCH/FS at 13 kbit/s Full-Rate, TCH/HS at 5.6 kbit/s Half-Rate, TCH/EFS at 12.2 kbit/s Enhanced Full-Rate) or data (TCH/F2.4, /F4.8, /F9.6). One TCH per active call (FS), or two calls sharing one physical channel (HS).
- **Control Channels (CCH):** Carry signaling and synchronization. Key channels:
  - **BCCH** (Broadcast Control Channel): Broadcasts cell identity, LAI, ARFCN list, access parameters. MS reads BCCH to camp on the cell.
  - **SCH** (Synchronization Channel): Carries BSIC (Base Station Identity Code) and TDMA frame number. MS uses SCH to achieve TDMA synchronization.
  - **FCCH** (Frequency Correction Channel): Unmodulated burst; MS uses to frequency-lock to the BTS carrier.
  - **RACH** (Random Access Channel): Uplink only; MS sends channel request here using slotted-ALOHA. First step in call/SMS origination.
  - **AGCH** (Access Grant Channel): Downlink only; BTS responds to RACH with an Immediate Assignment (assigning an SDCCH).
  - **PCH** (Paging Channel): Downlink; network pages MS for incoming call/SMS.
  - **SDCCH** (Stand-alone Dedicated Control Channel): Bidirectional; carries Layer 3 signaling (authentication, location update, SMS) before TCH assignment.
  - **SACCH** (Slow Associated Control Channel): Continuously associated with TCH or SDCCH; carries measurement reports (MS → BSC) and power control / timing advance commands (BSC → MS).
  - **FACCH** (Fast Associated Control Channel): Preempts the TCH (stealing bits from speech); carries urgent signaling (handover commands) requiring lower latency than SACCH.

3.3 Call Flow Diagrams

3.3.1 Mobile-Originated Call (MOC)

When a GSM subscriber initiates a call, the following sequence of events occurs. Each step involves specific Layer 3 signaling messages:

1. **Idle Mode → Channel Request:** The MS detects the user's call initiation (digit entry + send key). It transmits a **Channel Request** message on the RACH (slotted ALOHA). The BTS forwards it to the BSC.
2. **Immediate Assignment:** The BSC allocates an available SDCCH and instructs the BTS to send an **Immediate Assignment** message on the AGCH. The MS tunes to the assigned SDCCH.
3. **Authentication:** The MSC retrieves an (RAND, SRES, Kc) triplet from the VLR and sends **Authentication Request** (RAND) to the MS. The MS's SIM computes SRES = A3(Ki, RAND) and returns it. The MSC verifies. If SRES matches: authentication successful.
4. **Ciphering Mode:** The MSC sends **Ciphering Mode Command** (indicating which A5 algorithm and the Kc). The MS and BTS activate A5 encryption. From this point, all Um Layer 3 messages are encrypted.
5. **CM Service Request:** The MS sends **CM Service Request** (service type: MOC) on the SDCCH, identified by its TMSI.
6. **Setup Message:** The MS sends the **Setup** message (called party number, bearer capability) to the MSC via the BSC.
7. **TCH Assignment:** The BSC allocates a Traffic Channel (TCH/FS) and sends **Assignment Command** to the MS via the BTS. The MS moves from SDCCH to TCH.
8. **Call Proceeding → Alerting:** The MSC analyzes the dialed number, routes the call to the destination (PSTN or another MSC), and returns **Call Proceeding** then **Alerting** (ringing) messages to the MS.
9. **Connect:** When the called party answers, **Connect** is sent to the MS. The MS responds with **Connect Acknowledge**. The speech path is established end-to-end. Both parties can now speak.

3.3.2 Mobile-Terminated Call (MTC)

When an external party (PSTN or another mobile subscriber) calls a GSM subscriber, the network must first locate the called MS before routing the call:

10. **GMSC receives call:** The calling party dials the MSISDN. The call arrives at the **Gateway MSC (GMSC)** of the called subscriber's home operator.

11. **HLR Query:** The GMSC sends a **Send Routing Information (SRI)** MAP message to the called subscriber's HLR. The HLR looks up the subscriber's current VLR address.
12. **VLR Routing Request:** The HLR sends a **Provide Roaming Number (PRN)** MAP message to the currently serving VLR/MSC. The VLR allocates a temporary **Mobile Station Roaming Number (MSRN)** — a routable PSTN-format number.
13. **Call Routing:** The HLR returns the MSRN to the GMSC. The GMSC uses the MSRN to route the call directly to the serving MSC (Visited MSC) through the PSTN/SS7 network.
14. **Paging:** The serving MSC instructs all BSCs in the subscriber's registered **Location Area (LA)** to broadcast a **Paging Message** (containing the subscriber's TMSI) on the PCH of all cells in the LA.
15. **Paging Response:** The MS detects its TMSI on the PCH, requests a channel via RACH, receives an SDCCH assignment, and sends a **Paging Response** to the MSC.
16. **Authentication + Ciphering:** Same as MOC steps 3–4.
17. **TCH Assignment:** A TCH is allocated. The MS rings (alerting).
18. **Connect:** The subscriber answers. Connect messages are exchanged. The speech path is fully established.

3.4 Handover (Handoff)

Handover is the process of **transferring an active call from one radio channel (TCH) to another** while maintaining call continuity. In GSM, the decision to hand over is made by the **BSC or MSC** (not the MS), based on measurement reports collected from the MS and the serving BTS. GSM uses **hard handover** (Mobile-Assisted Handover, MAHO): the connection is briefly interrupted and re-established on the new channel, typically imperceptibly to the user (<50 ms).

Measurement Basis for Handover: The MS continuously measures the **RXLEV** (received signal level, mapped from RSSI) and **RXQUAL** (received signal quality, mapped from BER) of the serving cell, and measures RXLEV from up to 6 neighboring cells listed on the BCCH Allocation list (BA list). These measurements are averaged and reported to the BSC every 480 ms via the SACCH.

3.4.1 Handover Types

Handover Type	Between	Decision / Execution Authority
Intra-Cell HO	Two channels within the same cell (same BTS)	BSC decides and executes internally. Changes frequency or time slot only. Used to resolve interference or poor channel quality.
Inter-Cell, Intra-BSC HO	Two BTSs controlled by the same BSC	BSC decides and executes without MSC involvement. Fastest handover type. Uses internal BSC resources.
Inter-BSC, Intra-MSC HO	Two BSCs under the same MSC	BSC requests MSC to coordinate. MSC uses A interface to both BSCs. Slightly slower; MSC manages resource reservation on target BSC.
Inter-MSC HO	Two cells under different MSCs	Anchor MSC (call originator) coordinates with Relay MSC (target). Uses E interface SS7/MAP. Most complex: active call path rerouted through anchor.

⚠ **Handover Triggers**

RXLEV below threshold: Serving cell signal too weak (distance or obstruction). Most common trigger.

RXQUAL below threshold: Bit error rate too high (interference) despite acceptable signal level. Indicates co-channel or adjacent-channel interference.

TA (Timing Advance) too high: MS is too far from BTS (TA > 63 bits ≈ 35 km max). MS must hand over to closer BTS.

Power budget HO: A neighboring cell could serve the MS with less power (lower interference). Used to reduce network interference even before signal degradation.

4. GSM Protocols and Signaling

GSM's signaling architecture is organized as a **layered protocol stack** conforming to the OSI seven-layer reference model (reduced to three functional layers for the radio and terrestrial interfaces). Different protocol stacks apply to the Um (air) interface and to the terrestrial interfaces (Abis, A, and within the NSS). Understanding this layered structure is essential for analyzing GSM performance, troubleshooting faults, and understanding the evolution toward GPRS, EDGE, and beyond.

4.1 Protocol Stack on the Um Interface (MS ↔ BTS)

Layer 1 — Physical Layer (L1)

Layer 1 is responsible for **radio transmission and reception** on the Um interface. Key L1 specifications:

- **Modulation: Gaussian Minimum Shift Keying (GMSK)** with BT=0.3 (B = filter bandwidth, T = bit period). GMSK is a form of continuous-phase FSK with a Gaussian pre-modulation filter that reduces spectral sidelobes, improving adjacent-channel interference performance. Gross bit rate: **270.833 kbit/s** per carrier.
- **Burst structure:** Information is transmitted in **bursts** filling one time slot. Five burst types exist: **Normal Burst** (NB, 148 bits: 116 encrypted data bits + 26-bit training sequence + 3 tail bits + guard period); **Frequency Correction Burst (FB)**; **Synchronization Burst (SB)**; **Access Burst (AB)** (for RACH, short to accommodate unknown timing advance); **Dummy Burst (DB)** (filler when no information is transmitted).
- **Channel coding:** GSM uses **convolutional coding** (rate 1/2 for speech, with constraint length K=5) to add redundancy, enabling error detection and correction via **Viterbi decoding** at the receiver. Combined with **interleaving** over 8 consecutive bursts, this protects against burst errors caused by multipath fading.
- **Timing advance (TA):** Because propagation delay varies with distance (up to 35 km ≈ 117 µs one-way), the MS pre-advances its burst transmission by a TA value (0–63 bit periods, each 3.69 µs) commanded by the BTS, so that uplink bursts arrive at the BTS within their assigned time slot.

- **Power control:** MS transmit power is adjustable in 2 dB steps over a range of 5–33 dBm (MS class-dependent). This minimizes co-channel interference and conserves battery.

Layer 2 — Data Link Layer (L2)

Layer 2 provides **reliable, error-detected, and acknowledged data transfer** of Layer 3 signaling messages over the unreliable radio medium.

- **LAPDm (Link Access Protocol — Dm channel):** Used on the Um interface. Derived from the LAPD protocol of ISDN, adapted for the radio channel constraints (no line errors, but radio errors; no physical layer acknowledgment). Provides: framing (delimiting Layer 3 PDUs), error detection (CRC), sequencing, and retransmission (ARQ — Automatic Repeat Request) on dedicated channels (SDCCH, SACCH).
- **LAPDm on SACCH:** Unacknowledged mode on SACCH (measurement reports, power control): the channel is too lossy for ARQ; instead, the same data is transmitted repeatedly and the best copy is used.

Layer 3 — Network Layer (L3)

Layer 3 carries the substantive signaling logic of the GSM system. It is subdivided into three sub-layers, each with distinct responsibilities:

GSM Layer 3 Sub-Layers (Um Interface)

RR — Radio Resource Management: Lowest L3 sub-layer. Terminated at the BSC (not the MSC). Manages: cell selection and reselection, channel establishment and release, frequency hopping, handover execution, measurement reporting, timing advance, power control commands, ciphering mode command.

MM — Mobility Management: Terminated at the MSC/VLR. Manages: location updating (MS notifying the network of its new LA), IMSI attach/detach, authentication, TMSI reallocation, identity request.

CM — Connection Management: Highest L3 sub-layer. Terminated at the MSC. Contains three further entities: CC (Call Control — call setup/release), SMS-CP (SMS transfer), and SS (Supplementary Services — call forwarding activation, USSD).

4.2 Protocol Stack on the Abis Interface (BTS ↔ BSC)

The Abis interface carries both user traffic and signaling between BTS and BSC over standard **E1 (2.048 Mbit/s) PCM links** (or IP transport in modern iBTS/iAbis deployments):

- **L1 (Physical):** G.703/G.704 E1 framing. Each E1 carries 30 usable 64 kbit/s time slots (plus 2 for framing and signaling). Each GSM TCH/FS is compressed to **16 kbit/s** on Abis (4 GSM channels per E1 time slot) using TRAU sub-multiplexing.
- **L2 (Data Link): LAPD** (Link Access Procedure, D-channel — the full ISDN protocol, unlike the radio-adapted LAPDm). Provides reliable transport of signaling PDUs with ARQ.
- **L3 Signaling:** Carries two types of traffic: **Radio Signaling Link (RSL)** messages (BTS/BSC radio resource management, channel activation, measurement reports) and **Operations and Maintenance Link (OML)** messages (BTS configuration, software download, fault reporting), defined in 3GPP TS 48.058 and 52.021 respectively.

4.3 SS7 Signaling in the NSS

Within the NSS and between the BSS and NSS, GSM uses **Signaling System No. 7 (SS7)** as the underlying transport for all inter-element signaling. SS7 is the global standard for PSTN call control and telephony database signaling. Its layered structure as applied in GSM:

SS7 Layer / Protocol	GSM Application Protocol	Function in GSM
MTP Level 1 (Physical)	—	64 kbit/s or n×64 kbit/s E1 signaling links between SSPs/STPs
MTP Level 2 (Data Link)	—	Reliable L2 signaling link (FISU, LSSU, MSU frame types; HDLC-based)
MTP Level 3 (Network)	—	Signaling message routing and discrimination (Point Codes, SSN routing)
SCCP	Global Title Translation (GTT)	Connectionless and connection-oriented transport; translates MSISDN/IMSI to SS7 Point Code for HLR routing
TCAP	MAP / CAP transaction management	Transaction-based request/response (invoke/return result) for database queries (HLR, AuC)
MAP (Mobile Application Part)	Interfaces: B,C,D,E,F,G,H	GSM-specific: location updating, roaming, authentication, supplementary services, SMS routing (ETSI TS 129.002)
ISUP (ISDN User Part)	MSC ↔ PSTN / MSC ↔ MSC	Voice call setup/teardown on trunk circuits between MSC and PSTN or between MSCs
BSSAP / BSSMAP + DTAP	A Interface (BSC ↔ MSC)	BSSMAP: BSC–MSC resource management (handover, assignment). DTAP: transparent relay of MM/CC messages to/from MS

5. Measurement and Key Performance Indicators (KPIs)

Network measurement and Key Performance Indicator (KPI) analysis are fundamental to GSM network management, optimization, and troubleshooting. KPIs are computed from **raw measurement data** collected continuously by BTS, BSC, and MSC elements and transmitted to the OMC-R/OMC-S for aggregation and analysis. Two levels of measurement exist: **radio measurements** (performed by the MS and BTS on the Um interface) and **network-level counters** (call setup attempts, handover events, etc., counted by BSC and MSC). The following KPIs are the most critical for GSM radio network quality assessment.

5.1 Received Signal Strength Indicator (RSSI) / RxLev

5.1.1 Definition and Measurement

RSSI (Received Signal Strength Indicator) is the total received power at the antenna input, measured in **dBm**. In GSM, the measurement reported in signaling messages is called **RxLev** (Received Level), which is an integer in the range **0–63**, mapped from RSSI in dBm as:

> **RxLev = RSSI (dBm) + 110 [clipped to range 0–63]**
> Example: RSSI = −85 dBm → RxLev = 25. RxLev = 0 corresponds to RSSI ≤ −110 dBm (signal at or below noise floor). RxLev = 63 corresponds to RSSI ≥ −47 dBm (strong signal).

5.1.2 Measurement Procedure

The MS measures RXLEV of the serving cell and up to 6 neighboring BCCH carriers on TS0 of the respective ARFCN. These measurements are averaged over a 480 ms SACCH multiframe period and reported to the BSC in the **Measurement Report** message on the SACCH. The BTS simultaneously measures uplink RXLEV per MS burst. Both sets of measurements are used by the BSC for **handover decisions** and **power control commands**.

5.1.3 Engineering Significance

- **Cell selection and reselection:** An idle MS camps on the cell with the strongest BCCH (highest RxLev) that satisfies minimum signal threshold (RXLEV_MIN, broadcast on BCCH). RxLev < RXLEV_MIN triggers cell reselection.
- **Coverage assessment:** Drive test measurements of RSSI across the cell area produce coverage maps. Areas with RSSI below the minimum sensitivity threshold (≈ −102 dBm for 90% coverage probability) are identified as **coverage holes** requiring additional BTS or antenna adjustment.
- **Handover trigger:** If RxLev(serving) < RxLev(neighbor) + hysteresis (PBGT handover) or RxLev(serving) < L_RXLEV_DL_H threshold: BSC initiates handover toward the stronger neighbor.
- **Power control:** BSC commands MS/BTS to reduce power if RXLEV exceeds a target level (typically −70 to −75 dBm), minimizing co-channel interference.

5.2 Bit Error Rate (BER) and RxQual

5.2.1 Definition

BER (Bit Error Rate) is the fraction of received bits that are decoded incorrectly, expressed as a ratio or percentage. Two variants are distinguished in GSM:

- **BER_raw (BERraw):** The bit error rate measured **before** channel decoding (before error correction). Reflects the physical layer quality of the radio channel. Determined by SNR, multipath fading, and interference.
- **BER_residual (BERres):** The bit error rate measured **after** channel decoding (after convolutional code error correction). Reflects residual errors remaining after coding gain. Directly impacts speech quality perceived by the user.

In GSM signaling, BER is reported as the integer **RxQual** (Received Quality), mapped on a scale of **0–7**:

RxQual	BER Range (%)	Speech Quality	Engineering Interpretation
0	< 0.2%	Excellent	Ideal conditions; no perceptible artifacts
1	0.2 – 0.4%	Very Good	Clean speech; very low background noise
2	0.4 – 0.8%	Good	Minor artifacts; barely noticeable to user
3	0.8 – 1.6%	Acceptable	Some noise artifacts; acceptable for most users
4	1.6 – 3.2%	Marginal	Noticeable degradation; user may complain
5	3.2 – 6.4%	Poor	Significant noise; approaching call drop
6	6.4 – 12.8%	Bad	Heavy interference or deep fade; drop imminent
7	> 12.8%	Unusable	Link failure; call drop very likely

⚠ **Interference vs. Coverage Diagnosis**

A crucial diagnostic skill is distinguishing the **two principal causes** of poor RxQual: **(1) Coverage problem:** Low RxLev AND poor RxQual simultaneously → the MS is at the cell edge with insufficient signal power. Solution: reduce cell radius, add a new BTS, increase antenna height or tilt. (2) Interference problem: Low RxQual despite adequate RxLev → co-channel or adjacent-channel interference from another cell using the same ARFCN. Solution: frequency replanning, power reduction, antenna directivity adjustment.

5.3 Call Drop Rate (CDR)

5.3.1 Definition

Call Drop Rate (CDR) (also called Call Drop Ratio or Dropped Call Rate) is the percentage of calls that are **abnormally released** (disconnected without either party initiating hang-up) relative to the total number of calls that were established. It is computed at the BSC or MSC level from event counters:

> **CDR (%) = (Number of Dropped Calls / Number of Established Calls) × 100**
>
> A call is considered **dropped** when the radio link fails (RLL: Radio Link Failure, detected after N200 consecutive SACCH frames without acknowledgment), when handover fails and the call cannot recover, when the MS moves out of coverage, or when the BTS/BSC detects a hardware fault mid-call.

5.3.2 Target Values and Analysis

- **Industry benchmark:** CDR < 1.5% is considered **acceptable** for a well-operated GSM network. CDR > 3% indicates significant quality issues. Operators typically target CDR < 1% on the main carrier. Regulatory bodies in many countries set CDR < 2% as a mandatory KPI threshold.
- **Root cause analysis:** High CDR can result from: (a) Coverage holes (RXLEV < minimum sensitivity, no neighboring cell to hand over to); (b) Excessive interference causing Radio Link Failure (RXQUAL 6–7 for sustained period); (c) Failed handovers (target cell congested, incorrect neighbor list, timing issues); (d) Backhaul failures (Abis or A interface disconnection); (e) BTS hardware faults (TRX failure).
- **Spatial analysis:** CDR is correlated with cell identity and sector to isolate problematic cells. A cell with CDR significantly above the network average is a **worst-performing cell** requiring specific investigation (drive test, parameter audit, hardware check).

5.4 Call Setup Success Rate (CSSR)

5.4.1 Definition

Call Setup Success Rate (CSSR) measures the percentage of call setup attempts that result in a successfully established traffic channel (TCH), relative to the total number of setup attempts. It measures the network's ability to **grant service to requesting subscribers**:

> **CSSR (%) = (Successfully Established TCHs / Total Call Attempts) × 100**
>
> Note: Some operators decompose CSSR into sub-KPIs: **RACH Success Rate** (fraction of RACH requests receiving AGCH response), **SDCCH Seizure Success Rate** (fraction of SDCCH assignments completed), and **TCH Assignment Success Rate** (fraction of TCH assignments leading to a connected call). Each sub-KPI isolates a specific failure point in the setup chain.

5.4.2 Failure Causes and Thresholds

- **TCH congestion:** All available TCHs on the cell are occupied. New call requests receive a **congestion indication** and the call fails. Solution: add TRX hardware (more carriers), implement Half-Rate codec (doubles TCH capacity), add a new cell (cell splitting), or implement AMR (Adaptive Multi-Rate codec, compatible with higher network load).
- **SDCCH congestion:** All SDCCH slots occupied (heavy SMS traffic or location update storm). Configure more SDCCH slots by converting TCH capacity, or reduce the location update frequency parameter (T3212).
- **Radio failure during setup:** MS moves out of coverage or radio link degrades during SDCCH procedure, before TCH assignment. Reflected in poor RACH or SDCCH success rates.
- **Authentication failure:** SIM authentication fails (incorrect SRES response, AuC database inconsistency). Rare; usually operator provisioning issue.
- **Target:** CSSR > 95% is a typical minimum target. Well-optimized networks achieve CSSR > 98%. Regulatory thresholds typically require CSSR > 92–95%.

5.5 Additional GSM KPIs: Summary Table

KPI	Unit / Range	Good Target	Engineering Significance
RSSI / RxLev	dBm / 0–63	RxLev > 20 (≥−90 dBm)	Coverage; handover and PC trigger
RxQual	0–7 (0=best)	RxQual ≤ 3 (BER < 1.6%)	Interference diagnosis; speech quality proxy
CDR	Percentage (%)	< 1.5% (target < 1%)	Retained calls; user experience
CSSR	Percentage (%)	> 98% (min 95%)	Service accessibility; congestion detection
Handover Success Rate (HSR)	Percentage (%)	> 98%	Mobility management quality; neighbor list accuracy
Traffic (Erlang)	Erlang (Erl)	Varies by cell type	Dimensioning; congestion risk (Erlang B formula)
Timing Advance (TA)	0–63 (bit periods)	< 20 typical urban	Proxy for MS distance from BTS; macro-cell dimensioning
SD Success Rate (SDCCH)	Percentage (%)	> 99%	Signaling channel availability; SMS capacity
SQI (Speech Quality Index)	−30 to +30 (ITU-T)	> 20 (good speech)	Perceptual speech quality; combines RxLev + RxQual + DTX

5.6 KPI Collection Methods and Measurement Tools

5.6.1 Network-Side Statistics (OMC-R)

The OMC-R collects **performance measurement counters** from BSC and BTS at configurable intervals (typically 15-minute or 1-hour bins). These counters record every channel setup attempt, assignment, release (normal and abnormal), handover, and paging event. The OMC-R applies formulas to compute the KPIs above and presents them in tabular or

graphical (time-series, geo-referenced) reports. Threshold-based alarming alerts the network operations center when any KPI degrades beyond a set limit.

5.6.2 Drive Testing

Drive testing is the process of conducting measurements while physically traversing the network coverage area in a vehicle, using a **dedicated measurement MS** connected to a **TEMS** (Test Mobile System, e.g., TEMS Investigation) or similar drive test tool. The measurement MS logs all Layer 1, 2, and 3 parameters (RSSI, BER, RxLev, RxQual, serving cell ARFCN/BSIC, neighbor cell measurements, TA, power levels) along with GPS coordinates. The resulting logs are post-processed by tools such as **TEMS Discovery, Atoll, MapInfo**, or **Google Earth-based** visualization platforms to produce coverage maps, interference plots, and call quality geo-referenced overlays.

5.6.3 Key Drive Test Measurements

- **BCCH RSSI scan (idle mode):** Records the RSSI of all received BCCH carriers at each GPS location. Used to assess coverage quality and identify pilot pollution (too many cells with similar RSSI at one location, causing poor cell selection).
- **Dedicated mode measurement (active call):** Records RxLev, RxQual, TA, power control level, codec type, and neighbor cell levels during an active call. Identifies coverage holes, interference zones, and handover failure locations.
- **Call attempts:** The drive test MS makes periodic call attempts (e.g., every 60 seconds) to a test number. CSSR and CDR are measured directly from these attempts, correlated with GPS location to identify problem areas.

6. Chapter Summary: Key Concepts to Remember

Concept / Entity	Essential Definition / Key Point
GSM (2G)	First open-standard digital cellular system (1991). TDMA/FDMA access; 13 kbit/s RPE-LTP speech; A5 encryption; IMSI/SIM authentication; international roaming.
BSS (BSC + BTS)	Radio access subsystem. BTS: radio transceiver for one cell (1–6 TRX). BSC: controls 10–100+ BTSs; performs RR management, intra-BSC handover, power control.
Mobile Station (MS)	ME (handset, IMEI) + SIM (IMSI, Ki, A3/A8 algorithms). Ki never leaves SIM/AuC. TMSI used over air to protect IMSI.
MSC	Core switch: routes calls, manages mobility (location update), controls inter-MSC handover. GMSC: routes incoming calls by querying HLR.
HLR	Master subscriber database: IMSI, MSISDN, service profile, current VLR address. One per operator (logically). Queried by GMSC for MTC routing.
VLR	Local cache of subscriber data for MSs in current MSC area. Avoids repeated HLR queries. Assigns TMSI. Co-located with MSC.
AuC	Generates (RAND, SRES, Kc) triplets per subscriber using Ki + A3/A8. SRES verifies SIM. Kc encrypts Um interface with A5.
EIR	IMEI database: White (allowed), Grey (monitored), Black (barred). Checked by MSC at call setup via F interface.
FDMA layer (GSM-900)	200 kHz channel spacing; 124 ARFCNs (channels). 45 MHz UL/DL duplex offset. Each cell assigned a frequency subset.
TDMA layer	8 time slots per 200 kHz carrier; frame period 4.615 ms; slot duration 576.9 µs. 8 users per carrier. One physical channel = ARFCN + time slot number.

Concept / Entity	Essential Definition / Key Point
Frequency Reuse	N-cell cluster; D/R = √(3N). N=4 or N=7 typical. FH (slow frequency hopping) enables interference averaging → smaller N possible.
GMSK Modulation	Gaussian Minimum Shift Keying; BT=0.3; 270.833 kbit/s gross bit rate; constant envelope (efficient RF amplification); compact spectrum.
GSM L3 Protocols	RR (radio resource, terminated BSC): channel mgmt, HO. MM (mobility mgmt, terminated VLR): location update, authentication. CM (connection mgmt, terminated MSC): call control, SMS, SS.
SS7 / MAP	NSS inter-element signaling. MTP1/2/3 + SCCP + TCAP + MAP. MAP carries: location updates (D interface), routing info (C), inter-MSC HO (E), IMEI check (F). ISUP for voice trunks.
MOC (call origination)	RACH → AGCH → SDCCH → Authentication → Ciphering → Setup → TCH Assignment → Alerting → Connect. 9-step signaling sequence.
MTC (call termination)	GMSC queries HLR (SRI) → HLR queries VLR (PRN) → VLR returns MSRN → GMSC routes to serving MSC → MSC pages LA → MS responds → Auth + Cipher → TCH → Connect.
Handover (MAHO)	Mobile-Assisted Hard Handover. Decision by BSC/MSC from MS measurement reports. Types: intra-cell, inter-cell/intra-BSC, inter-BSC/intra-MSC, inter-MSC.
RxLev / RSSI	Received signal level. RxLev = RSSI(dBm) + 110; range 0–63. Used for coverage assessment, handover triggers, power control.
RxQual / BER	Received quality. BER mapped to RxQual 0–7 (0=best, <0.2%; 7=worst, >12.8%). Distinguishes interference (high RxQual, adequate RxLev) from coverage (both low).

Concept / Entity	Essential Definition / Key Point
CDR	Call Drop Rate. Abnormal releases / established calls × 100%. Target < 1.5%. Caused by coverage holes, interference, failed HO, hardware faults.
CSSR	Call Setup Success Rate. Successful TCH assignments / attempts × 100%. Target > 98%. Fails due to TCH/SDCCH congestion, radio failure during setup.

Looking Ahead: Chapter 4

Chapter 4 will build directly on the GSM architecture studied here to introduce GPRS (General Packet Radio Service) and EDGE (Enhanced Data rates for GSM Evolution) — the '2.5G' and '2.75G' packet-data overlays on the GSM network. We will examine how GPRS introduces packet switching into the BSS and NSS through the SGSN and GGSN, how radio blocks and coding schemes (CS-1 to CS-4, MCS-1 to MCS-9 for EDGE) work, and how these systems bridge the gap between circuit-switched 2G voice and fully packet-based 3G/4G networks.

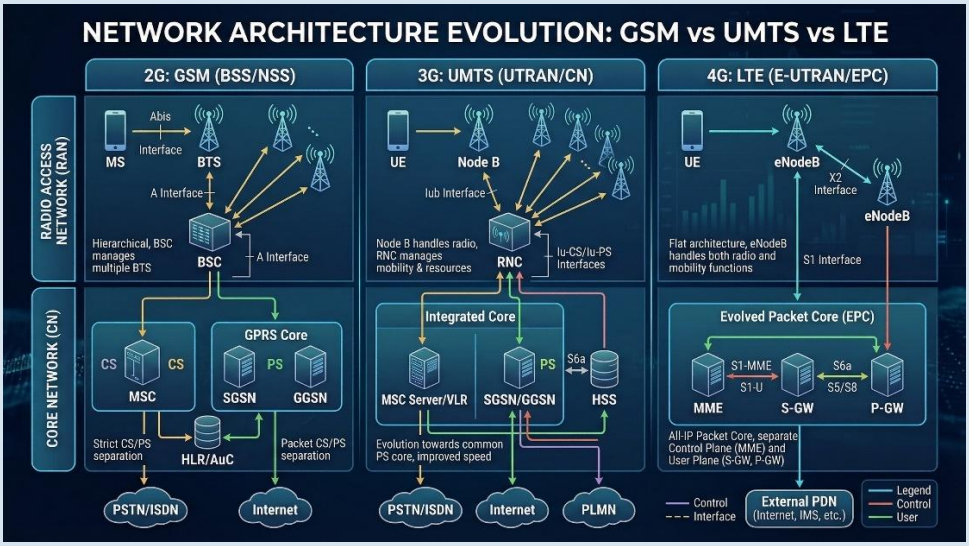
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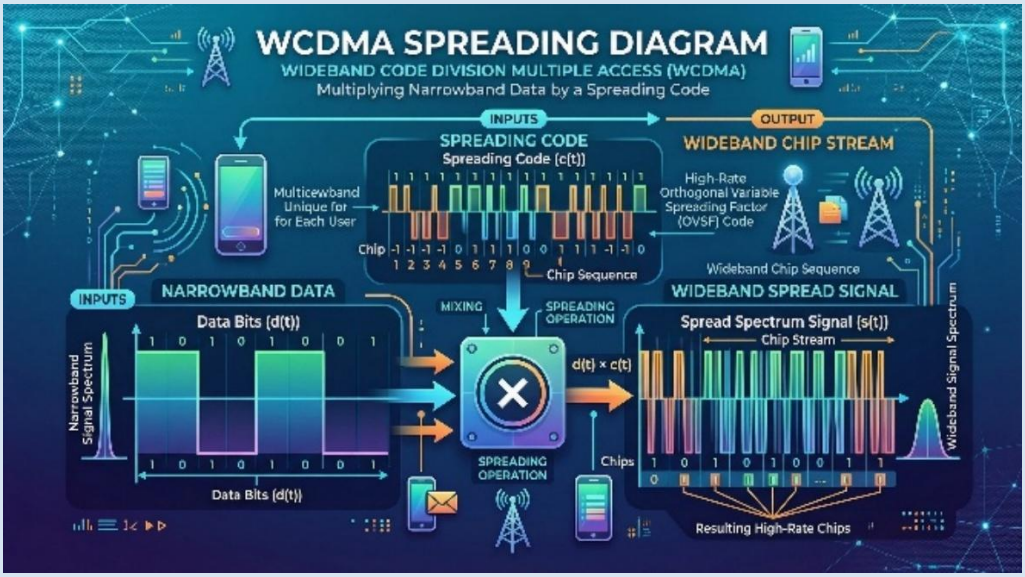
CHAPTER 4

The New Generations of Digital Telephony:

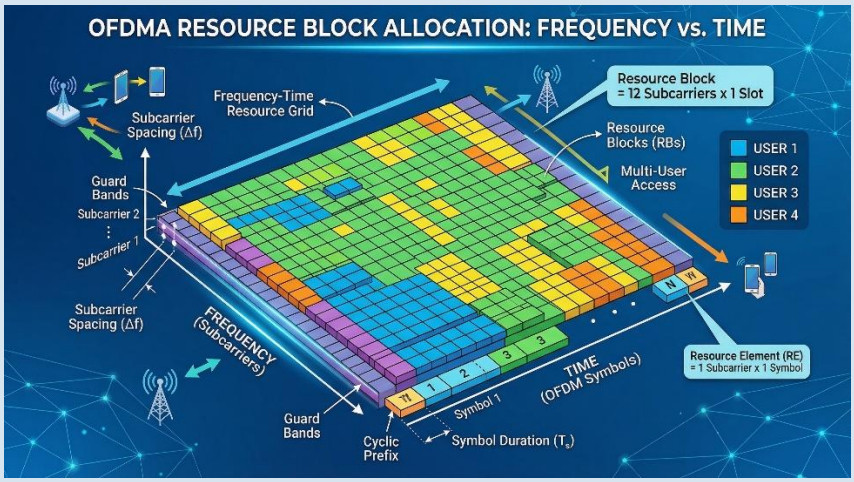
3G / UMTS — 3.5G / HSPA — 4G / LTE — 5G and Beyond



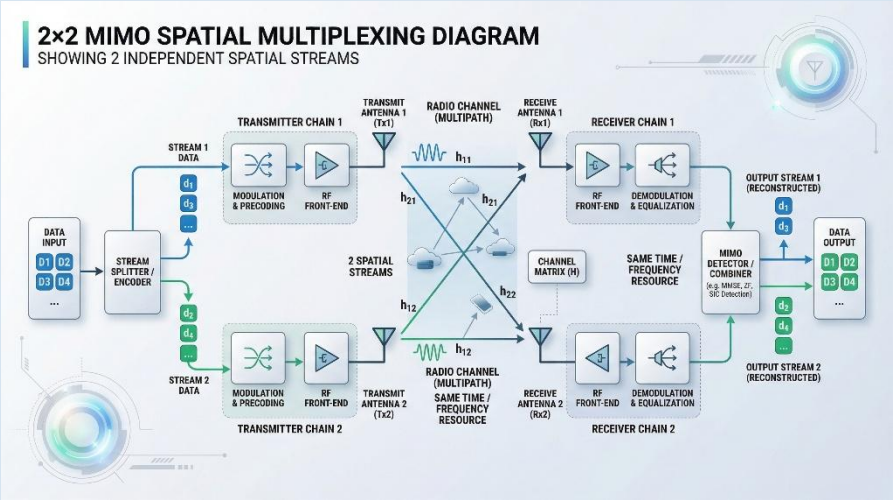
(1) A side-by-side network architecture comparison: GSM BSS/NSS vs. UMTS UTRAN/CN vs. LTE E-UTRAN/EPC, showing how the radio and core network elements evolved across generations.



(2) A WCDMA spreading diagram: narrowband data × spreading code → wideband chip stream.



(3) An OFDMA subcarrier grid: frequency vs. time resource block allocation.



(4) A 2×2 MIMO antenna diagram showing spatial multiplexing streams.

1. Transition from GSM (2G) to Third Generation (3G)

1.1 Pedagogical Objectives

Upon successful completion of this chapter, the student will be able to:

- **Identify** the principal technical and commercial limitations of GSM that motivated the development of third and fourth generation systems.
- **Describe** the UMTS/3G network architecture (UE, NodeB, RNC, CN-CS, CN-PS) and the role of each element.
- **Explain** the W-CDMA air interface principle (spreading, orthogonality, soft capacity) and contrast it with GSM's TDMA/FDMA scheme.

- **Describe** the HSPA (HSDPA/HSUPA) enhancements that constituted 3.5G, including higher-order modulation and fast scheduling.
- **Explain** the LTE/4G motivation, key radio technologies (OFDMA, MIMO), and the simplified E-UTRAN + EPC all-IP architecture.
- **Present** the objectives and key architectural concepts of 5G NR (New Radio) and articulate the evolutionary path from 2G to 5G.

1.2 Limitations of GSM: The Driver for Evolution

GSM was an extraordinary engineering achievement that served billions of subscribers for over three decades. However, as the commercial Internet expanded rapidly in the late 1990s and mobile data services began to emerge, several **fundamental technical limitations** of the GSM/2G system became increasingly constraining:

1.2.1 Extremely Low Data Rates

- **Circuit-Switched Data (CSD):** GSM's original data service used a single 9.6 kbit/s circuit-switched channel — equivalent to the speed of a 1990s analogue modem. Establishing a data session consumed a full dedicated TCH for the entire call duration, whether or not data was actively being transferred, at the same per-minute tariff as voice.
- **GPRS (2.5G):** The General Packet Radio Service (GPRS, 3GPP Release 97) introduced **packet-switched data** over the GSM radio interface, allowing up to 4 time slots to be combined for data (multi-slot operation), yielding a theoretical maximum of **171.2 kbit/s** (4 slots × 21.4 kbit/s with CS-4 coding scheme) but a typical practical throughput of 20–40 kbit/s. This enabled "always-on" connectivity but remained far below the requirements for multimedia services.
- **EDGE (2.75G):** Enhanced Data rates for GSM Evolution (GPRS Release 99+) introduced **8-PSK modulation** (replacing GMSK) to achieve three bits per symbol instead of one, and new coding schemes (MCS-1 to MCS-9), raising the theoretical maximum to **384 kbit/s** and practical throughput to 100–150 kbit/s. While a significant improvement, EDGE remained a bandwidth-limited overlay on the GSM radio interface, not a fundamental redesign.

1.2.2 Circuit-Switched Architecture

The GSM core network was inherited from the PSTN: it was fundamentally **circuit-switched**, designed for constant-bit-rate voice calls. Every data session required dedicating fixed network resources (time slots, trunks, MSC switching capacity) for the entire duration, even during idle periods. This was **spectral and economic waste** for bursty internet data traffic, which by nature is highly variable in rate and has very different QoS requirements from voice.

1.2.3 Limited Spectral Efficiency

The TDMA/FDMA scheme of GSM allocated each user a fixed time slot and fixed frequency channel. Users who momentarily had nothing to transmit (silent periods in conversation, pauses in data transfer) still occupied their allocated slot, wasting spectrum. Additionally, the **narrowband 200 kHz channel** fundamentally limited the raw bit rate per carrier, regardless of modulation improvements.

Summary: GSM Data Rate Evolution vs. Requirements

GSM CSD: 9.6 kbit/s. Sufficient for: SMS, basic WAP. Insufficient for: web browsing, email attachments.

GPRS (2.5G): ~20–40 kbit/s typical. Sufficient for: simple web, MMS. Insufficient for: video streaming, VoIP.

EDGE (2.75G): ~100–150 kbit/s typical. Sufficient for: enhanced web, audio streaming. Insufficient for: video calls, fast download.

ITU IMT-2000 (3G target): 144 kbit/s (vehicular), 384 kbit/s (pedestrian), 2 Mbit/s (indoor). This required a fundamentally new radio access technology.

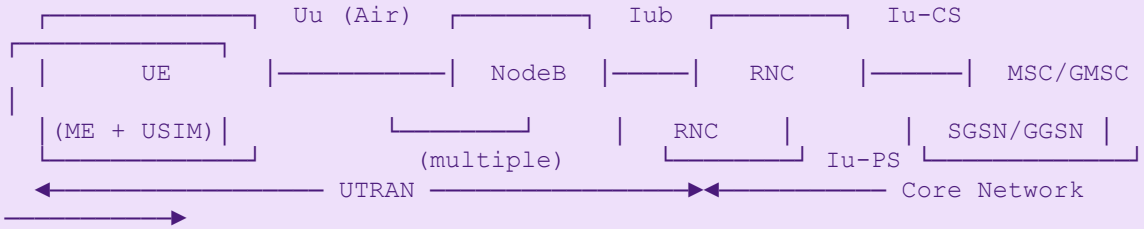
2. Third Generation (3G): UMTS and W-CDMA

The **Universal Mobile Telecommunications System (UMTS)** is the dominant 3G standard developed by 3GPP (3rd Generation Partnership Project) to meet the ITU's **IMT-2000** requirements. First commercially deployed in 2001 (Japan, NTT DoCoMo), UMTS introduces a completely new **Radio Access Network (UTRAN)** based on **W-CDMA** (Wideband Code Division Multiple Access), while retaining a largely backward-compatible core network. This duality allowed operators to reuse GSM MSC/HLR infrastructure for voice while introducing a new packet-switched domain for data.

2.1 UMTS Network Architecture

The UMTS architecture is organized into three major domains, replacing the GSM BSS/NSS division:

UMTS Architecture Overview



UTRAN = UMTS Terrestrial Radio Access Network. Core Network (CN) is split into CS domain (voice) and PS domain (data/internet).

2.1.1 User Equipment (UE)

The **User Equipment (UE)** is the subscriber's terminal, equivalent to the GSM MS. It consists of:

- **Mobile Equipment (ME):** The handset hardware with W-CDMA radio transceiver (5 MHz channels), RAKE receiver, display, and multimedia processing capability (video codec, camera interface). Each ME is identified by an **IMEI** (or IMEISV for software versioning).
- **USIM (Universal Subscriber Identity Module):** An upgraded SIM (on a UICC — Universal Integrated Circuit Card) that stores the IMSI, long-term key **K** (replacing GSM's Ki), and implements the UMTS security algorithms (**MILENAGE**: f1–f5),

which generate **quadruplets** (RAND, XRES, CK, IK) rather than GSM triplets. The USIM also supports **mutual authentication**: the network authenticates the UE **and** the UE authenticates the network, preventing the false base station (IMSI catcher) attack possible in GSM.

2.1.2 NodeB (Base Station)

The **NodeB** is the UMTS radio base station, analogous to the GSM BTS. It serves one or more cells (typically 3 sectors). Key functions:

- **W-CDMA physical layer:** Spreading and despreading of user data using orthogonal codes (OVSF — Orthogonal Variable Spreading Factor codes), QPSK modulation (uplink), QPSK or 16-QAM modulation (downlink with HSDPA), and RAKE receiver combining for multipath diversity.
- **Inner loop power control:** Adjusts transmit power at **1500 Hz** (once per slot, every 0.667 ms) in 1 dB steps, based on SIR (Signal-to-Interference Ratio) measurements. This **fast closed-loop power control** is the critical mechanism that counters the near-far problem in CDMA (a nearby user's signal overwhelming a distant user's signal).
- **Macro-diversity (soft handover support):** Simultaneously transmits to and receives from a UE on multiple NodeBs during soft handover, with the RNC combining the received signals (Maximum Ratio Combining).

2.1.3 Radio Network Controller (RNC)

The **Radio Network Controller (RNC)** is the intelligence of the UTRAN, analogous to the GSM BSC but significantly more capable. Each RNC manages multiple NodeBs and is connected to the Core Network via the **Iu interface**:

- **Iu-CS:** Connects RNC to the MSC/GMSC (Circuit-Switched core). Carries voice calls (AMR-coded at 12.2 kbit/s) and associated signaling (RANAP protocol over SS7/ATM).
- **Iu-PS:** Connects RNC to the SGSN (Packet-Switched core). Carries IP data traffic and GPRS/UMTS packet data signaling (GTP-U tunnels).

RNC functions include: radio resource management (code allocation, congestion control), outer loop power control, soft handover combining (Serving-RNC / Drift-RNC architecture via the **Iur interface** between RNCs), header compression (PDCP layer), and admission control.

2.1.4 UMTS Core Network (CN)

The UMTS CN is split into two parallel domains that reflect the dual nature of 3G services (voice and broadband data):

- **Circuit-Switched (CS) Domain:** Reuses and enhances the GSM MSC, GMSC, HLR, VLR, AuC, and EIR infrastructure. Handles AMR voice calls using traditional circuit switching over ATM or TDM trunks. The **MSC Server** (in Release 4+) separates call control (signaling, handled by MSC Server) from bearer switching (handled by a Media Gateway, MGW), moving toward a softer switching model.
- **Packet-Switched (PS) Domain:** New elements introduced in GPRS (Release 97) and enhanced for UMTS:
  - **SGSN (Serving GPRS Support Node):** Routes packet data to/from the UE, manages PS mobility (routing area updates), and handles authentication and ciphering for the PS domain. Connected to RNC via Iu-PS.
  - **GGSN (Gateway GPRS Support Node):** The gateway between the PS domain and external packet data networks (internet, corporate intranets). Assigns IP addresses to UEs, enforces QoS policies, and provides the anchor point for PS mobility.
  - **HLR/HSS:** The HLR (enhanced to HSS — Home Subscriber Server in later releases) stores both CS and PS subscription data. Queried by SGSN for PS authentication and service profiles.

2.2 The W-CDMA Air Interface

The replacement of GSM's TDMA/FDMA air interface with **W-CDMA (Wideband Code Division Multiple Access)** is the most fundamental technical change from 2G to 3G. W-CDMA uses a **5 MHz wideband channel** (25 times wider than GSM's 200 kHz) and separates users not by time slots or frequency sub-bands, but by unique **orthogonal spreading codes**.

2.2.1 The Spreading Principle

In W-CDMA, each user's data stream (at a rate of 7.5 kbit/s to 2 Mbit/s, depending on the service) is **spread** by multiplying it bit-by-bit with a high-rate **chip sequence** (a pseudo-random binary code) at the fixed **chip rate of 3.84 Mchips/s**. The ratio of chip rate to data rate is the **Spreading Factor (SF)**, also called the processing gain:

Spreading Factor SF = Rchip / Rdata = 3.84 Mchips/s / Rdata

SF ranges from 4 to 512 (in steps of powers of 2). High data rate services (e.g., 384 kbit/s data) use SF=8. Voice (12.2 kbit/s AMR) uses SF=128. The processing gain (PG = $10 \times \log_{10}(SF)$ in dB) is the SNR improvement from spreading — it allows the signal to be transmitted **below the noise floor**, recoverable only by someone with the correct spreading code.

Example: Data rate = 12.2 kbit/s (AMR voice). SF = 3,840,000 / 12,200 ≈ 315 → rounded to SF=256. Processing gain = $10 \times \log_{10}(256) \approx 24$ dB. This 24 dB gain allows the transmitted signal to be buried 24 dB below the wideband noise and interference, yet recovered by the RAKE receiver using the correct code.

2.2.2 OVSF Codes and Orthogonality

UMTS uses **Orthogonal Variable Spreading Factor (OVSF) codes**, derived from a **Hadamard code tree**. Codes at the same level of the tree are mutually orthogonal, meaning a properly synchronized receiver can perfectly separate all users' signals even though they occupy the same 5 MHz bandwidth simultaneously. The OVSF tree allows flexible allocation: a high-SF (low data rate) code can coexist with a low-SF (high data rate) code at a different branch of the tree, subject to the constraint that parent codes cannot be simultaneously allocated.

2.2.3 CDMA Key Properties vs. TDMA/FDMA

Property	GSM TDMA/FDMA	UMTS W-CDMA
Channel bandwidth	200 kHz (narrow)	5 MHz (wideband)
User separation	Time slot + frequency assignment	Orthogonal spreading code
Frequency planning	Complex: N=4 or N=7 cluster required	Universal frequency reuse possible (N=1): same 5 MHz in all cells
Multipath handling	Equalization (limited, narrowband)	RAKE receiver: exploits multipath as diversity (constructive combining)

Property	GSM TDMA/FDMA	UMTS W-CDMA
Handover	Hard handover (break-before-make)	Soft handover (make-before-break): UE connects to 2–6 NodeBs simultaneously
Capacity limit	Hard limit: fixed number of time slots	Soft limit: interference-limited; more users → more interference → gradual quality decrease
Power control	Slow (power control commands every few hundred ms)	Fast closed-loop: 1500 Hz (every 0.667 ms slot); critical to CDMA near-far mitigation
Peak data rate (Rel99)	~9.6 kbit/s CSD; 40–150 kbit/s GPRS/EDGE	384 kbit/s (outdoor); 2 Mbit/s (indoor, low mobility)

2.2.4 UMTS Frame Structure

The UMTS radio frame has a duration of **10 ms** (identical to GSM for synchronization compatibility reasons). Each frame is divided into **15 slots** of 0.667 ms each. This slot is the basic unit of power control (one power control command per slot). Three adjacent slots form a **subframe of 2 ms** (the scheduling unit for HSDPA, introduced in Section 3). The physical channels within each slot carry different combinations of data bits, pilot bits, and transport format indicators.

2.3 Key Improvements of UMTS over GSM

- **Multimedia services:** UMTS introduced **video telephony** (H.324M, 64 kbit/s on CS domain), **streaming audio/video** (RTSP over PS domain), **multimedia messaging (MMS)**, and broadband internet access — services fundamentally impossible at GSM data rates.
- **Packet-data integration:** The dual CS/PS core network enabled operators to offer **"always-on" IP connectivity** (persistent IP address from GGSN) alongside traditional voice, without the resource waste of circuit-switched data.
- **Improved spectral efficiency:** W-CDMA's universal frequency reuse (N=1), soft handover gain, and RAKE receiver multipath combining yield significantly better

spectral efficiency than GSM TDMA/FDMA under high-load conditions, particularly in dense urban deployments.

- **Enhanced security:** UMTS MILENAGE algorithm provides **mutual authentication** (the network also proves its identity to the UE, preventing IMSI catchers), and introduces 128-bit **confidentiality key (CK)** and 128-bit **integrity key (IK)** for air interface encryption and signaling integrity protection.
- **QoS classes:** UMTS defines four QoS traffic classes: **Conversational** (voice, video call — strict delay), **Streaming** (audio/video streaming — variable delay tolerance), **Interactive** (web browsing, email — best effort with round-trip constraint), and **Background** (file download, email sync — delay-tolerant best effort). This QoS framework enables the network to prioritize voice and real-time services over background downloads.

3. Evolution within 3G: 3.5G — HSPA and HSPA+

Although UMTS Release 99 provided a substantial improvement over GSM data rates, practical deployed peak rates were often limited to 128–384 kbit/s due to channel conditions and load. The **High Speed Packet Access (HSPA)** family, introduced in 3GPP Release 5 (HSDPA, 2002) and Release 6 (HSUPA, 2004), and further evolved to **HSPA+** in Release 7 and 8 (2007–2008), constitutes the **3.5G** evolution layer: a set of enhancements to the UMTS physical and MAC layers that dramatically increased data throughput with minimal changes to the network architecture.

3.1 HSDPA — High Speed Downlink Packet Access (3GPP Release 5)

HSDPA achieves a theoretical peak downlink rate of **14.4 Mbit/s** (Category 10 UE) through four key mechanisms applied to a new shared channel, the **High Speed Downlink Shared Channel (HS-DSCH)**:

3.1.1 Adaptive Modulation and Coding (AMC)

Rather than using a fixed modulation and coding rate for all users regardless of channel quality (as in W-CDMA Release 99), HSDPA uses **Adaptive Modulation and Coding**

(AMC): the NodeB continuously selects the most efficient combination of modulation order and channel coding rate based on the UE's reported channel quality:

- **QPSK:** 2 bits/symbol. Used when channel quality is poor (UE far from NodeB, heavy interference). More robust coding (higher redundancy).
- **16-QAM:** 4 bits/symbol. Used under moderate channel conditions. Doubles spectral efficiency vs. QPSK.
- **64-QAM:** 6 bits/symbol (introduced in HSPA+, Release 7). Used only in very favorable channel conditions (UE close to NodeB, high SINR). Triples spectral efficiency vs. QPSK.

The UE reports its **Channel Quality Indicator (CQI)** — an index from 0 to 30 that encodes the recommended transport block size and modulation — every **2 ms** (one TTI) on the uplink HS-DPCCH channel. The NodeB's MAC-hs scheduler reads these CQI reports and dynamically selects the highest modulation/coding combination each UE can reliably decode, maximizing cell throughput.

3.1.2 Fast Hybrid ARQ (HARQ)

HSDPA introduces **Hybrid Automatic Repeat reQuest (HARQ)** at the **NodeB MAC layer** (not at the RNC as in Release 99). When a transport block is incorrectly received (CRC error), the UE sends a NACK (Negative ACKnowledgment) on the HS-DPCCH. The NodeB retransmits the same block, and critically, the UE uses **Chase Combining** or **Incremental Redundancy** to **soft-combine** the erroneous and retransmitted versions — extracting more information from the accumulated received bits. The combined block can be successfully decoded even when neither individual transmission was decodable alone.

HARQ round-trip time: The HARQ RTT in HSDPA is **8 × 2 ms = 16 ms** (a fixed 8-process Stop-and-Wait HARQ mechanism). This is dramatically faster than the Release 99 RLC retransmission delay of 50–300 ms, significantly improving effective throughput in error-prone channel conditions.

3.1.3 NodeB-Based Fast Scheduling

In HSDPA, the scheduling function is moved from the RNC to the **NodeB MAC-hs entity**, operating on a **2 ms Transmission Time Interval (TTI)** granularity (compared to 10 ms or 20 ms in Release 99). This proximity to the radio interface allows the scheduler to react to **fast channel variations (fast fading)** on a sub-frame timescale.

Two principal scheduling strategies are used:

- **Round Robin:** Each UE is served in turn, regardless of channel quality. Ensures fairness but suboptimal cell throughput.
- **Proportional Fair:** Schedules the UE whose current CQI is highest relative to its own long-term average. This exploits **multi-user diversity**: different users experience peak channel quality at different moments; by always scheduling the currently "best" user, average cell throughput is substantially increased while maintaining long-term fairness.

3.2 HSUPA — High Speed Uplink Packet Access (3GPP Release 6)

HSUPA (formally named **Enhanced Dedicated Channel, E-DCH**) applies analogous enhancements to the **uplink** direction, achieving a theoretical peak uplink rate of **5.76 Mbit/s**:

- **Short TTI (2 ms):** Reduces uplink scheduling and HARQ feedback latency compared to the 10 ms TTI of Release 99 DCH.
- **NodeB-coordinated uplink scheduling:** The NodeB grants UEs permission to transmit at specified rates (absolute or relative grants), managing uplink interference (the **noise rise** above thermal noise) to stay within a target level (typically 3–6 dB rise). This prevents any single UE from overloading the cell with high-power uplink transmissions.
- **Uplink HARQ:** Same Hybrid ARQ principle as HSDPA, with NodeB soft-combining of retransmitted uplink transport blocks.

3.3 HSPA+ (Release 7/8/9) and Peak Rate Progression

Release	Standard	DL Peak Rate	UL Peak Rate	Key Technology
Rel. 99	UMTS W-CDMA	384 kbit/s	128 kbit/s	QPSK, SF codes
Rel. 5	HSDPA (3.5G)	14.4 Mbit/s	384 kbit/s	AMC, HARQ, NodeB sched.
Rel. 6	HSUPA (3.5G)	14.4 Mbit/s	5.76 Mbit/s	E-DCH, UL HARQ, grants

Release	Standard	DL Peak Rate	UL Peak Rate	Key Technology
Rel. 7	HSPA+ (3.75G)	28 Mbit/s	11.5 Mbit/s	64-QAM DL, 16-QAM UL, MIMO 2×2
Rel. 8	HSPA+ DC	42 Mbit/s	11.5 Mbit/s	Dual Carrier HSDPA (2×5 MHz)
Rel. 9	HSPA+ DC-MIMO	84 Mbit/s	23 Mbit/s	Dual Carrier + 2×2 MIMO combined

HSPA+ — through the combination of 64-QAM modulation, 2×2 MIMO spatial multiplexing, and dual-carrier aggregation (bonding two adjacent 5 MHz carriers) — achieved peak downlink rates of 42–84 Mbit/s, approaching the performance of early LTE deployments. This allowed operators to **significantly extend the commercial life of UMTS infrastructure** before full LTE deployment, protecting their 3G capital investments.

4. Fourth Generation (4G): LTE and LTE-Advanced

4.1 Motivation: Why a “Long-Term Evolution”?

By 2004, 3GPP recognized that incremental enhancements to W-CDMA would be insufficient to meet the exploding demand for mobile broadband driven by smartphones (iPhone launched 2007), video streaming, cloud services, and machine-to-machine communication. A radical re-design was required. The **Long Term Evolution (LTE)** project (3GPP Release 8, 2008) and its successor **LTE-Advanced** (Release 10, 2011, meeting ITU IMT-Advanced requirements) set aggressive targets:

- **Peak downlink:** 100 Mbit/s (LTE Release 8); 1 Gbit/s (LTE-A Release 10 with Carrier Aggregation).
- **Peak uplink:** 50 Mbit/s (LTE Release 8); 500 Mbit/s (LTE-A).

- **Latency:** User-plane latency < 5 ms (one-way, unloaded); control-plane connection setup time < 100 ms.
- **Spectrum flexibility:** Channel bandwidths from 1.4 to 20 MHz (scalable: 1.4, 3, 5, 10, 15, 20 MHz).
- **All-IP architecture:** Elimination of the circuit-switched domain entirely. Voice is handled as **VoLTE (Voice over LTE)** — a VoIP service with IMS (IP Multimedia Subsystem) QoS guarantees, equivalent in quality to circuit-switched voice.
- **Spectral efficiency:** 3× improvement over HSPA in downlink; 2× in uplink.

4.2 Key Technology 1: OFDMA — Orthogonal Frequency Division Multiple Access

4.2.1 OFDM Principle

LTE replaces W-CDMA with **OFDM (Orthogonal Frequency Division Multiplexing)** on the downlink. OFDM divides the available channel bandwidth into a large number of closely spaced, narrowband **subcarriers**, each modulated at a low symbol rate:

> **Subcarrier spacing: $\Delta f = 15$ kHz | Subcarrier symbol duration: $T_s = 1/\Delta f = 66.7$ µs**
>
> **Subcarriers per bandwidth:** 1.4 MHz: 72 subcarriers (6 RBs). 5 MHz: 300 (25 RBs). 10 MHz: 600 (50 RBs). 20 MHz: 1200 (100 RBs). Each Resource Block = 12 subcarriers × 1 slot (0.5 ms) = 12×7 = 84 resource elements.
>
> **Cyclic Prefix (CP):** A guard interval of 4.7 µs (normal CP) or 16.7 µs (extended CP for high-delay-spread environments) is prepended to each OFDM symbol. The CP must be longer than the channel's delay spread. Eliminates inter-symbol interference (ISI) without complex time-domain equalization.

4.2.2 OFDM Advantages for Mobile Channels

- **Robustness to multipath:** Because each subcarrier's symbol duration (66.7 µs) is much longer than the channel delay spread (typically 1–10 µs in urban environments), multipath echoes arriving within the CP are absorbed without causing ISI. A single-tap frequency-domain equalizer per subcarrier suffices, regardless of channel complexity.
- **Efficient spectrum use:** OFDM subcarriers are mathematically orthogonal ($\int s_i(t) \cdot s_j(t)\, dt = 0$ for $i \neq j$), meaning they can be placed with zero guard band between

them while remaining perfectly separable at the receiver. Spectral efficiency approaches the Nyquist theoretical limit.

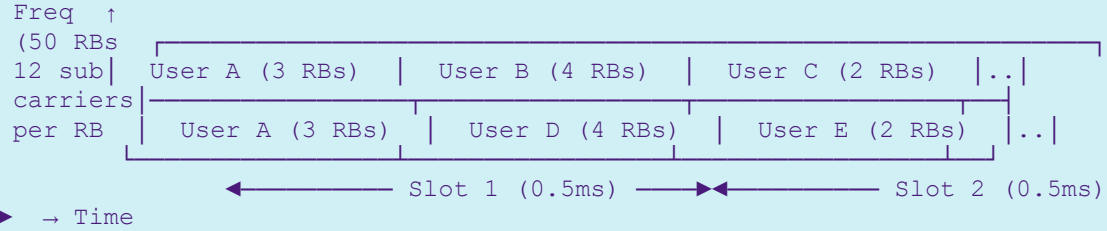
- **Flexible resource allocation:** Each **Resource Block (RB)** — 12 subcarriers × 0.5 ms — can be independently assigned to different users, enabling fine-grained time-frequency resource management by the scheduler.
- **PAPR issue:** OFDM's main disadvantage is a high **Peak-to-Average Power Ratio (PAPR)**, requiring linear power amplifiers with significant power back-off. This is acceptable for the eNB (mains-powered) but problematic for battery-powered UE uplink transmitters.

4.2.3 SC-FDMA for the Uplink

To avoid the PAPR problem on the uplink (where battery power is critical), LTE uses **SC-FDMA (Single Carrier – Frequency Division Multiple Access)** instead of OFDMA. SC-FDMA applies an additional DFT (Discrete Fourier Transform) pre-coding step before the OFDM modulator, resulting in a signal with single-carrier PAPR characteristics while preserving the same frequency-domain multiple access structure and equalization simplicity as OFDMA. SC-FDMA is sometimes called **DFT-spread OFDM (DFT-S-OFDM)**.

4.2.4 OFDMA Resource Grid

LTE Resource Grid (10 MHz channel, 1 subframe = 1 ms)



Each Resource Block = 12 subcarriers (180 kHz) × 1 slot (0.5 ms). Each user is assigned a set of RBs per subframe (1 ms). The scheduler allocates RBs based on CQI reports (channel quality) and QoS requirements, achieving optimal multi-user diversity.

4.3 Key Technology 2: MIMO — Multiple Input Multiple Output Antennas

MIMO is the second fundamental enabling technology of LTE, enabling substantial increases in both peak data rate (through **spatial multiplexing**) and link reliability (through **transmit diversity**) by exploiting multiple physical antenna elements at both the transmitter and receiver.

4.3.1 MIMO Transmission Modes

A MIMO system with N_T transmit antennas and N_R receive antennas is written **$N_T \times N_R$ MIMO**. LTE defines several transmission modes (TM1–TM9), each optimizing for different channel conditions and UE capabilities:

MIMO Mode	Configuration	Mechanism	Best Used When
Transmit Diversity (TM2)	2 or 4 Tx antennas	SFBC (Space-Frequency Block Coding): same data on all Tx with coding for diversity	Poor channel conditions (cell edge); low SNR; high mobility
Open Loop Spatial Multiplexing (TM3)	2×2 or 4×4	Parallel data streams on independent spatial channels; no CSI feedback	Good channel conditions; UE moving fast (CSI feedback unreliable)
Closed Loop Spatial Multiplexing (TM4)	2×2 or 4×4	Precoding matrix selected by eNB using PMI (Precoding Matrix Indicator) from UE feedback	Good channel, low mobility; UE reports PMI+RI+CQI feedback
MU-MIMO (TM5)	Multiple UEs on same RBs	Spatial separation of multiple users on same time-frequency resource using beamforming	Dense deployment; multiple UEs with spatially separated channels
Beamforming (TM7/8)	Up to 8 Tx antennas	Concentrate radiated energy toward target UE using phase-	Cell edge; interference mitigation; high spatial isolation needed

MIMO Mode	Configuration	Mechanism	Best Used When
		weighted antenna array	

4.3.2 Spatial Multiplexing: Capacity Gain

In **spatial multiplexing** mode, the transmitter simultaneously sends **multiple independent data streams (layers)** from different antennas. The multiple receive antennas, combined with **MIMO detection algorithms** (MMSE, ML, or successive interference cancellation), can separate these streams because they have passed through **different spatial propagation channels** and thus carry different multipath signatures.

> **MIMO Capacity (Foschini-Gans): $C = \Sigma \log_2(1 + \lambda i \times \rho/NT)$ [bits/s/Hz]**
>
> Where λ_i are the eigenvalues of the channel matrix $H \cdot H^H$ (proportional to the independent spatial channel gains) and ρ is the average SNR per receive antenna. For a 4×4 MIMO system with rich scattering (rank-4 channel), capacity approaches **4×** the SISO capacity at the same SNR. LTE Release 8 supports up to **4 spatial layers** (4×4 MIMO); LTE-A supports up to **8 layers (8×8 MIMO)**. 5G NR supports up to 32 layers in Massive MIMO configurations.

4.3.3 UE Feedback for MIMO

Efficient closed-loop MIMO requires the UE to report three feedback quantities on the PUCCH (Physical Uplink Control Channel):

- **CQI (Channel Quality Indicator):** Recommended modulation and code rate for each codeword (independent data stream), corresponding to a target 10% block error rate.
- **PMI (Precoding Matrix Indicator):** Index into a codebook of predefined precoding matrices that best aligns the transmitted streams with the UE's channel eigenvectors, maximizing received SNR.
- **RI (Rank Indicator):** The recommended number of spatial layers (rank 1 to 4/8). The rank is limited by min(N_T, N_R) and by the channel's spatial rank (number of significant eigenvalues). At cell edge or in low-scattering environments, rank = 1 (transmit diversity); in rich scattering near the eNB, rank = 2, 3, or 4.

4.4 LTE Network Architecture: E-UTRAN and EPC

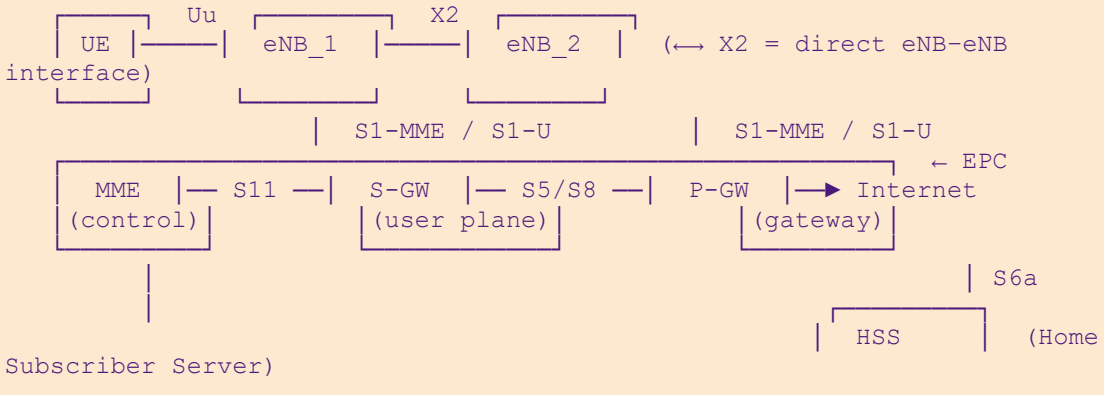
LTE introduces a fundamentally simplified **flat, all-IP architecture** that eliminates the hierarchical structure of UMTS (NodeB → RNC → Core) and the CS domain entirely. The result is lower latency, reduced infrastructure cost, and a fully IP-native service model.

4.4.1 E-UTRAN: Evolved Universal Terrestrial Radio Access Network

The E-UTRAN consists solely of **eNBs (evolved NodeBs)**. All radio resource management functions previously housed in the RNC are **distributed to the eNB itself**. There is no separate RNC:

- **eNB functions:** Radio resource management (scheduling, handover decisions, admission control, interference coordination), OFDMA/SC-FDMA physical layer, HARQ management, IP header compression (PDCP), user-plane ciphering and integrity, and measurement reporting to SON (Self-Organizing Network) functions.
- **X2 Interface:** A direct IP-based interface between adjacent eNBs (X2-C for control plane, X2-U for user plane). Used for: **X2-based handover** (the most common handover type: source eNB directly requests the target eNB to prepare resources, then notifies the MME after handover completion — faster than S1-based handover via MME); **interference coordination** (coordinated multi-point, ICIC — Inter-Cell Interference Coordination, exchanging load and resource usage information).
- **S1 Interface:** Connects each eNB to the EPC. Splits into S1-MME (control plane: S1-AP protocol) and S1-U (user plane: GTP-U tunnels over UDP/IP).

LTE Architecture Diagram



4.4.2 EPC: Evolved Packet Core

The **Evolved Packet Core (EPC)** is a fully packet-switched, IP-based core network. It replaces the entire GSM/UMTS NSS (MSC, SGSN, GGSN) with four principal elements:

- **MME (Mobility Management Entity):** The **control-plane anchor** for the UE. Handles: NAS (Non-Access Stratum) signaling — UE attach/detach, EPS bearer setup, authentication and security context establishment (using AKA — Authentication and Key Agreement), tracking area management (TA list assignment), paging, and selection of S-GW/P-GW. The MME carries **no user-plane traffic**: a clean control/user-plane separation.
- **S-GW (Serving Gateway):** The **user-plane anchor** for the E-UTRAN. Routes user-plane IP packets between eNB and P-GW via GTP-U tunnels. Serves as the mobility anchor within the E-UTRAN: during intra-LTE handover, the S-GW maintains the GTP tunnel toward the P-GW while the eNB-side tunnel endpoint changes. Also buffers downlink packets during paging (UE is in idle mode).
- **P-GW (PDN Gateway):** The **gateway to external PDN (Packet Data Networks)**: the internet, corporate VPNs, or IMS. Functions: IP address allocation (via DHCPv4/v6 or static), policy enforcement (PCEF — Policy and Charging Enforcement Function, applying rules from PCRF), deep packet inspection, traffic shaping, and bearer-level QoS enforcement. The P-GW is the anchor for inter-technology mobility (handover between LTE and 3G/2G preserves the IP address at the P-GW).
- **HSS (Home Subscriber Server):** Evolved form of the GSM HLR, enhanced for LTE. Stores EPS subscription data, PDN connection profiles, and authentication vectors. Connected to MME via the S6a interface (Diameter protocol, replacing SS7/MAP).

☑ Key Architectural Simplifications in LTE vs. UMTS

No RNC: All radio resource management in the eNB itself. Eliminates RNC bottleneck; enables distributed processing.

No CS Domain: Voice is VoLTE over IMS — fully IP, same pipe as data. Eliminates dual CS/PS architecture complexity.

Flat user plane: UE → eNB → S-GW → P-GW → Internet. Fewer nodes → lower latency (< 5 ms radio hop vs. 50+ ms in UMTS).

Control/User plane separation: MME handles only signaling (control); S-GW/P-GW handle only data (user). Scales independently — foundation of 5G SBA architecture.

4.5 LTE-Advanced: Carrier Aggregation and Massive MIMO

4.5.1 Carrier Aggregation (CA)

LTE-A Release 10 introduced **Carrier Aggregation (CA)**: the simultaneous use of up to **5 Component Carriers (CC)**, each up to 20 MHz wide, for a combined bandwidth of up to **100 MHz**. This achieves the LTE-A 1 Gbit/s peak rate target (100 MHz × 8 bits/s/Hz spectral efficiency × (layers/streams)). CA can aggregate carriers in the same band (intra-band contiguous CA), in the same band but non-contiguous (intra-band non-contiguous CA), or in different bands (inter-band CA — e.g., 800 MHz + 2600 MHz simultaneously for coverage + capacity).

4.5.2 Enhanced ICIC and CoMP

- **eICIC (enhanced Inter-Cell Interference Coordination):** In heterogeneous networks (macrocells overlaid with small cells), Almost Blank Subframes (ABS) allow macro cells to mute their transmission on selected subframes so that small cell UEs can be scheduled interference-free.
- **CoMP (Coordinated Multi-Point):** Multiple geographically separated eNBs simultaneously transmit to (or receive from) the same UE, using precoding coordination. Joint transmission CoMP can significantly boost cell-edge throughput (up to 2× improvement demonstrated in field trials).

5. 5G New Radio (NR) and Beyond: An Overview

The **Fifth Generation (5G)**, standardized as **5G New Radio (NR)** in 3GPP Release 15 (2018) and continuously evolving through Releases 16 (2020), 17 (2022), 18 (2024), and 19+, represents not merely an incremental performance improvement over LTE, but a **fundamental architectural and paradigm shift**: 5G is designed to be a universal wireless platform serving a broad and heterogeneous ecosystem of use cases far beyond mobile telephony — encompassing critical industrial automation, massive IoT, immersive media, and ultra-reliable emergency communications.

5.1 ITU IMT-2020: Three 5G Use Case Families

The ITU-R M.2083 framework defines three complementary service categories for 5G, each with distinct and challenging technical requirements:

Use Case Family	Acronym	Representative Applications and Key Requirements
Enhanced Mobile Broadband	eMBB	8K video streaming, VR/AR headsets, stadium/hotspot ultra-dense: peak DL > 20 Gbit/s; peak UL > 10 Gbit/s; user experience rate 100 Mbit/s.
Ultra-Reliable Low-Latency Communication	URLLC	Industrial automation, autonomous vehicles, remote surgery, smart grid: latency ≤ 1 ms (user-plane); reliability ≥ 99.9999% (6 nines); packet error rate ≤ 10^{-5}.
Massive Machine-Type Communication	mMTC	Smart cities, precision agriculture, industrial sensor networks, logistics tracking: connection density up to 10^6 devices/km²; ultra-low power (10+ year battery); low data rate (bps to kbps per device).

5.2 Key 5G Technical Innovations

5.2.1 Millimeter-Wave Spectrum (FR2)

5G NR is the first cellular standard to extensively use **millimeter-wave (mmWave) frequencies**: the **FR2 (Frequency Range 2)** band spans 24.25–52.6 GHz (with up to 400 MHz bandwidth per carrier and up to **8 aggregated carriers = 3.2 GHz**). This extreme bandwidth enables peak data rates exceeding 10 Gbit/s per cell. The challenges of mmWave (high path loss, poor obstacle penetration, rain fade) are addressed by **analog and hybrid beamforming** using large antenna arrays (Massive MIMO — see below). FR1 (sub-6 GHz: 450 MHz–6 GHz) provides coverage and macro-layer capacity comparable to LTE.

5.2.2 Massive MIMO and Beamforming

5G NR eNBs (called **gNBs**) deploy **Massive MIMO** antenna arrays with 32, 64, 128, or even 512 antenna elements at the base station. Using **3D beamforming** (both horizontal azimuth and vertical elevation steering — called **FD-MIMO or Full-Dimension MIMO**), the gNB can:

- **Form narrow, precise beams** directed at individual UEs, dramatically concentrating signal energy and minimizing interference to neighboring cells and users.
- **Simultaneously serve tens of users** on the same time-frequency resource (massive MU-MIMO), each with a spatially isolated beam, multiplying cell capacity proportionally.
- **Tilt beams vertically** to serve UEs on different floors of high-rise buildings (3D coverage), impossible with 2D antenna arrays.

5.2.3 Flexible Numerology and Slot Structure

Unlike LTE's fixed 15 kHz subcarrier spacing, 5G NR supports **multiple numerologies (μ = 0 to 4)**, each doubling the subcarrier spacing:

μ	SCS (Δf)	Slot Duration	Slots/Subframe	Application
0	15 kHz	1 ms	1	Sub-6 GHz eMBB; backward compat.
1	30 kHz	0.5 ms	2	Sub-6 GHz eMBB; reduced latency
2	60 kHz	0.25 ms	4	FR1 URLLC; FR2 mmWave eMBB
3	120 kHz	0.125 ms	8	mmWave (FR2) eMBB; sub-ms latency

μ	SCS (Δf)	Slot Duration	Slots/Subframe	Application
4	240 kHz	0.0625 ms	16	mmWave reference signal (future)

Multiple numerologies can coexist within the same carrier (mixed numerology), serving different traffic types: a URLLC slot with μ=3 (0.125 ms duration) can be pre-empted into an eMBB frame with μ=0 without disturbing the overall subframe structure.

5.2.4 5G NR Waveform: CP-OFDM and DFT-S-OFDM

5G NR uses **CP-OFDM** (Cyclic Prefix OFDM, same as LTE) on both downlink **and** uplink (unlike LTE's SC-FDMA uplink). For coverage-limited scenarios (cell edge, mmWave link budgets), **DFT-S-OFDM** (LTE's SC-FDMA) is also supported on the uplink as an optional waveform, retaining the low-PAPR advantage for power-limited UEs.

5.3 The 5G Core Network: Service-Based Architecture (SBA)

The 5G core network (5GC), defined in 3GPP Release 15, introduces a **Service-Based Architecture (SBA)** that replaces the point-to-point interface model of EPC with a **microservices-based** design inspired by cloud-native software architecture. Each core network function exposes a **REST API over HTTP/2** (using JSON or Protocol Buffers), and any authorized function can discover and consume any other function's services through a **Network Repository Function (NRF)** — similar to a service registry in cloud computing.

Key 5GC network functions (NFs):

- **AMF (Access and Mobility Management Function):** Replaces MME. NAS signaling, UE registration, mobility, paging, authentication coordination.
- **SMF (Session Management Function):** Manages PDU sessions (IP session setup, QoS enforcement commands, UPF selection). Separated from mobility management.
- **UPF (User Plane Function):** Replaces S-GW+P-GW user plane. Routes user data packets; enforces QoS; interfaces with external data networks. Can be deployed at edge (MEC — Multi-access Edge Computing) for ultra-low latency.
- **PCF (Policy Control Function):** Replaces PCRF. Provides policy rules to SMF and AMF.
- **AUSF (Authentication Server Function):** Replaces AuC; implements 5G AKA and EAP-AKA' authentication.

- **UDM (Unified Data Management):** Replaces HSS. Manages subscriber data; generates authentication credentials.
- **NSSF (Network Slice Selection Function):** Selects the appropriate network slice instance for the UE's requested service.

5G Network Slicing

One of 5G's most significant innovations is **network slicing**: the ability to create multiple independent, logically isolated virtual networks (**slices**) over the same physical 5G infrastructure, each customized for a specific use case. A URLLC slice for autonomous vehicle control (strict latency, high reliability, small data) coexists on the same physical radio and core infrastructure as an eMBB slice for video streaming (high throughput, tolerable latency) and an mMTC slice for IoT sensors (massive connections, low power, low rate). Each slice has independently configured AMF/SMF/UPF instances with its own QoS policies, resource budgets, and security parameters.

5.4 Beyond 5G: Towards 6G (Release 19+ and IMT-2030)

Research and early standardization for **6G (IMT-2030)** is underway in academia and at ITU-R, with commercial deployments expected around 2030. Current 6G vision elements include:

- **Terahertz (THz) spectrum (0.1–1 THz):** Potentially hundreds of GHz of bandwidth per link, enabling Tbit/s peak rates. Extreme propagation challenges (molecular absorption, very short range) — primarily relevant for ultra-dense indoor and device-to-device scenarios.
- **Reconfigurable Intelligent Surfaces (RIS):** Large panels of passive electromagnetic reflectors that can be programmed to redirect and focus radio waves, effectively controlling the propagation environment — a fundamental departure from treating the radio channel as a fixed, uncontrollable medium.
- **AI-native air interface:** Machine learning integrated into physical layer design (AI-based channel estimation, AI-based MIMO detection, AI-driven resource management), replacing hand-crafted algorithms with learned functions.
- **Integrated terrestrial and non-terrestrial networks (NTN):** Seamless integration of LEO/MEO satellite constellations, HAPS (High-Altitude Platform Stations), and

terrestrial 6G cells into a unified network fabric, providing true global coverage continuity.

- **Integrated sensing and communication (ISAC):** The 6G radio simultaneously serves as a high-resolution radar/sensor (for environmental monitoring, gesture recognition, and positioning) and a communication link, sharing spectrum and hardware.

6. Generation Comparison: From GSM to 5G

Parameter	GSM (2G)	GPRS/EDGE (2.5/2.75G)	UMTS Rel99 (3G)	HSPA+ (3.75G)	LTE / LTE-A (4G)	5G NR (5G)
Air Interface	TDMA/FDMA GMSK	GMSK / 8-PSK	W-CDMA QPSK	W-CDMA + AMC 64-QAM	OFDMA/SC-FDMA up to 256-QAM	CP-OFDM up to 1024-QAM
Channel BW	200 kHz	200 kHz	5 MHz	5/10 MHz	1.4–20 MHz (CA: 100 MHz)	50/100/400 MHz (CA: 3.2 GHz)
Peak DL	9.6 kbit/s	40–384 kbit/s	384 kbit/s–2 Mbit/s	84 Mbit/s	150 Mbit/s / 1 Gbit/s	20 Gbit/s (FR2)
Peak UL	9.6 kbit/s	40–150 kbit/s	128–384 kbit/s	23 Mbit/s	50 Mbit/s / 500 Mbit/s	10 Gbit/s (FR2)
Latency (RTT)	~500 ms	300–500 ms	50–100 ms	20–50 ms	10–30 ms / < 5 ms	< 1 ms (URLLC)
Core Architecture	CS only (MSC/HLR)	CS + PS (SGSN/GGSN)	CS + PS (MSC + SGSN)	CS + PS (enhanced)	All-IP EPC (MME/S-GW/P-GW)	SBA 5GC (AMF/SMF/UPF)
MIMO	None	None	None (Rel99)	2×2 MIMO (Rel7+)	4×4 / 8×8 MIMO	Massive MIMO (up to 512 elements)
Spectrum	850/900/1800/1900 MHz	Same as GSM	850/900/2100 MHz	Same as UMTS	700 MHz–2.6 GHz (LTE); CA multi-band	Sub-6 GHz (FR1) + 24–52 GHz (FR2)
3GPP Release	Rel. 97–98	Rel. 97–99	Rel. 99	Rel. 5–9	Rel. 8–13	Rel. 15–19+

7. Chapter Summary: Key Concepts to Remember

Concept / Term	Essential Definition / Key Point
GSM Limitations	9.6 kbit/s CSD; circuit-switched core wastes spectrum on bursty data; 200 kHz TDMA/FDMA limits raw bit rate; no multimedia support.
GPRS / EDGE	GPRS (Rel.97): multi-slot packet data, ~40 kbit/s typical. EDGE (Rel.99): 8-PSK modulation, ~150 kbit/s. PS overlay on GSM; introduces SGSN/GGSN.
UMTS / W-CDMA	3G standard (3GPP Rel.99). 5 MHz channel. Users separated by OVSF spreading codes. Chip rate 3.84 Mchips/s. RAKE receiver. Soft handover. Fast power control 1500 Hz.
Spreading Factor (SF)	SF = chip rate / data rate. Range 4–512. Processing gain = $10 \times \log_{10}(SF)$ dB. High SF = robust low-rate channel; Low SF = high-rate channel.
Soft Handover	Make-before-break: UE connected to 2–6 NodeBs simultaneously. RNC combines signals (MRC). Reduces handover drop risk; uses more network resources.
UMTS Core: CS+PS	CS domain: MSC(Server)+MGW for voice. PS domain: SGSN (local routing, auth) + GGSN (internet gateway). HLR/HSS serves both.
USIM / MILENAGE	USIM replaces SIM. MILENAGE f1–f5 algorithms generate quadruplets (RAND, XRES, CK, IK). Mutual authentication: network also proves identity to UE.
HSDPA (Rel.5)	HS-DSCH shared channel. Peak 14.4 Mbit/s. Adaptive Modulation and Coding (QPSK/16-QAM/64-QAM). NodeB fast scheduler (2 ms TTI). HARQ with soft combining.

Concept / Term	Essential Definition / Key Point
HSUPA (Rel.6)	E-DCH channel. Peak 5.76 Mbit/s UL. NodeB-coordinated uplink scheduling. 2 ms TTI. HARQ. Manages uplink noise rise.
AMC (Adaptive Modulation/Coding)	Selects modulation + coding rate per TTI based on CQI report from UE. High SINR: 64-QAM; Low SINR: QPSK. Maximizes cell throughput across all channel conditions.
OFDMA (LTE DL)	15 kHz subcarrier spacing. Cyclic prefix eliminates ISI. Resource Block = 12 subcarriers × 0.5 ms. Scheduler assigns RBs per user per subframe. Optimal multi-user diversity.
SC-FDMA (LTE UL)	DFT-spread OFDM. Same frequency-domain resource structure as OFDMA but single-carrier PAPR. Critical for battery-powered UE uplink power efficiency.
MIMO Spatial Multiplexing	N_T × N_R MIMO: up to min(N_T,N_R) independent data streams. Capacity scales linearly with rank. UE reports CQI+PMI+RI feedback. LTE: up to 8×8; 5G: up to Massive MIMO (512 elements).
LTE Architecture (EPC)	No RNC; all RRM at eNB. X2 interface: direct eNB–eNB. EPC: MME (control plane only) + S-GW (user-plane anchor) + P-GW (PDN gateway). HSS replaces HLR. Diameter replaces SS7/MAP.
Carrier Aggregation (CA)	LTE-A Rel.10: up to 5 component carriers (CC) × 20 MHz = 100 MHz. Intra-band contiguous/non-contiguous; inter-band. Achieves 1 Gbit/s peak.
5G NR Use Cases	eMBB (>20 Gbit/s peak, >100 Mbit/s user exp.). URLLC (≤1 ms latency, 99.9999% reliability). mMTC (10^6 devices/km², ultra-low power).

Concept / Term	Essential Definition / Key Point
5G NR Numerology	Multiple subcarrier spacings: 15/30/60/120/240 kHz (μ=0–4). Slot duration halves per step. Mixed numerology supports eMBB+URLLC on same carrier.
Massive MIMO (5G)	32–512 antenna elements at gNB. 3D beamforming (azimuth + elevation). Simultaneous multi-user beams on same TF resource. Key to 5G spectral efficiency.
5GC Service-Based Architecture	Microservices: AMF, SMF, UPF, PCF, AUSF, UDM, NSSF communicate via REST/HTTP2. NRF for service discovery. Enables cloud-native deployment and network slicing.
Network Slicing	Multiple isolated virtual networks on same physical 5G infrastructure. Each slice has dedicated AMF/SMF/UPF and custom QoS/security. Serves eMBB, URLLC, mMTC simultaneously.

Looking Ahead: Module Conclusion

This chapter completes the four-chapter arc of the Telephony module: from the analog circuit-switched PSTN of Chapter 1, through the transmission media that carry all signals (Chapter 2), to GSM 2G digital cellular (Chapter 3), and finally to the modern broadband mobile generations (Chapter 4). The progression from 9.6 kbit/s GSM CSD to 20 Gbit/s 5G NR — achieved over just 25 years — illustrates the extraordinary pace of innovation driven by the interplay of semiconductor technology (Moore's Law enabling complex signal processing), information theory (approaching Shannon limits through advanced modulation and coding), and systems architecture (from monolithic circuit-switched hierarchies to cloud-native microservice platforms). The principles mastered in this module — multiple access, channel coding, modulation, network architecture, quality measurement — form the analytical foundation for all advanced work in wireless communications, network engineering, and telecommunications systems design.

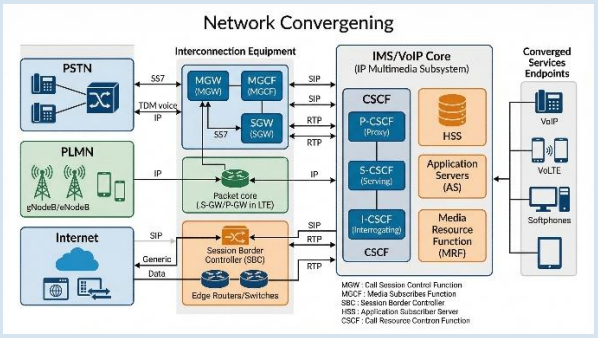
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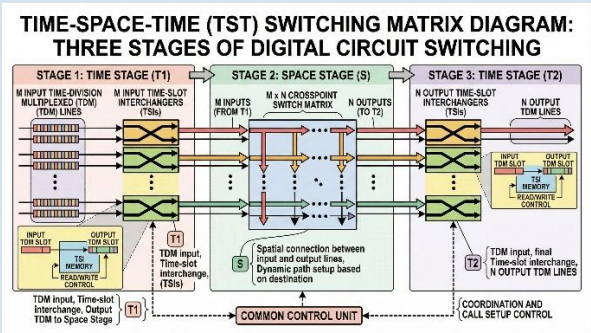
CHAPTER 5

Telephony Interconnection Equipment:

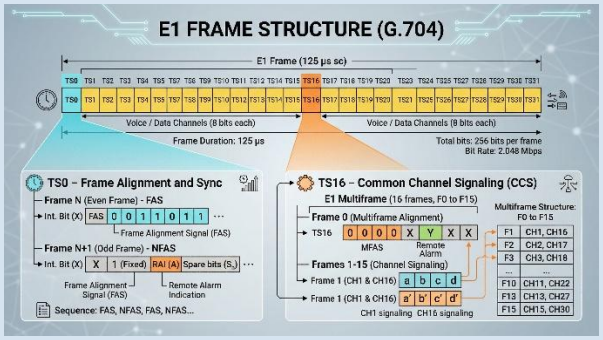
Switches, Routers, Gateways, and Interfaces



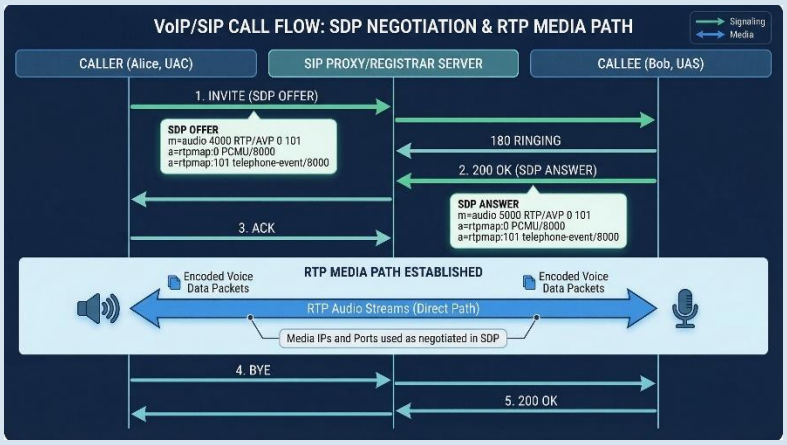
(1) A network convergence diagram showing PSTN, PLMN, and Internet converging through interconnection equipment (switches, gateways, routers) toward an IMS/VoIP core.



(2) A Time-Space-Time (TST) switching matrix diagram showing the three stages of digital circuit switching.



(3) An E1 frame structure diagram (32 time slots, TS0 framing, TS16 signaling).



(4) A VoIP/SIP call flow diagram showing SDP negotiation and RTP media path.

1. Introduction to Telephony Interconnection

1.1 Pedagogical Objectives

Upon successful completion of this chapter, the student will be able to:

- **Explain** why interconnection equipment is indispensable in heterogeneous telecommunications networks that mix analog, digital circuit-switched, and IP-based technologies.
- **Describe** the operation of digital telephone switches, including the principles of Time-Division and Space-Division switching, and the TST/STS switching fabric architecture.
- **Explain** the role of IP routers in carrying voice traffic (VoIP) and the QoS mechanisms that ensure voice quality in packet networks.
- **Define** the functions of Media Gateways (MGW) and Signaling Gateways (SGW), and explain how they enable interworking between circuit-switched networks and VoIP/IP networks.
- **Describe** the standard physical and protocol interfaces used for trunk interconnection: E1/T1 PRI, SS7/ISUP, SIP trunks, and Gigabit Ethernet/fiber optic transport.
- **Apply** a comparative framework to select the appropriate interconnection equipment for a given network scenario.

1.2 The Interconnection Imperative

Modern telecommunications networks are not monolithic: they are a complex ecosystem of **heterogeneous technologies** that have evolved over more than a century and now coexist within and between operator networks. A telephone call originating from a mobile subscriber (GSM or LTE) may traverse a mobile core network, cross an international border via SS7 signaling and TDM trunks, pass through a Voice over IP (VoIP) platform, and terminate on a legacy analog POTS telephone — all in a matter of seconds.

This heterogeneity creates a fundamental engineering challenge: each network segment uses different **physical transmission media**, different **signaling protocols**, different **codec and voice encoding formats**, and different **addressing and numbering schemes**. Without specialized **interconnection equipment** — switches, routers, gateways, and the interfaces that connect them — communication between these segments would be impossible.

The evolution of interconnection technology has itself driven the transformation of the entire telecommunications industry: the transition from analog to digital switching (1970s–1980s), from digital TDM networks to VoIP over IP (1990s–2000s), and from monolithic hardware appliances to virtualized, cloud-hosted network functions (2010s–2020s) each required new interconnection paradigms. Understanding these equipment categories and interfaces is therefore essential for any telecommunications engineer.

The Three Interconnection Dimensions

Technology Interworking: Connecting networks that use fundamentally different communication paradigms: analog circuit switching (PSTN), digital circuit switching (ISDN/SS7), and packet switching (IP/SIP/RTP).

Protocol Translation: Converting between signaling protocols: SS7/ISUP ↔ SIP, SS7/MAP ↔ Diameter, H.323 ↔ SIP, QSIG ↔ SIP.

Media Format Conversion: Transcoding between voice representations: analog waveforms ↔ PCM (G.711) ↔ compressed codecs (G.729, AMR) ↔ IP/RTP packets.

2. Digital Switches (Commutateurs Numériques)

The **digital telephone switch** (also called a digital exchange or digital switching system) is the central processing element of the circuit-switched telephone network. It replaced the electromechanical crossbar switches of the analog era with a combination of **digital time-division multiplexing (TDM)** and **digital space-division switching** using high-speed semiconductor logic, enabling far higher capacity, reliability, and feature richness at lower cost per circuit.

2.1 The Role of the Digital Switch

A digital telephone switch performs the fundamental function of **establishing, maintaining, and releasing connections** between any two subscriber lines or trunk circuits in its domain. In the digital domain, this means connecting the **time slot** carrying the PCM samples of subscriber A with the time slot that will carry those samples to subscriber B — and simultaneously connecting B's voice in the opposite direction. The switch must perform this connection for thousands or tens of thousands of simultaneous calls.

Beyond basic switching, modern digital exchanges perform: **concentration** (multiplexing many low-traffic subscriber lines onto fewer trunk circuits), **expansion** (the reverse on the output side), **call processing** (analyzing dialed digits, applying routing tables, generating signaling), **tone generation** (dial tone, busy, ring-back), and **CDR generation** (Call Detail Records for billing).

2.2 Time-Division Switching (TDS)

Time-Division Switching is the mechanism that routes PCM voice samples from one time slot (TS) of an incoming TDM highway to a different time slot on an outgoing TDM highway. It exploits the fact that in a TDM system, each voice channel occupies a brief, recurring time slot, and **the switch only needs to read a sample from one slot and write it into another** to create a virtual circuit.

2.2.1 Time-Stage (T-Stage) Switching

A Time-Stage switch is implemented as a **speech memory (SM)** and a **connection memory (CM)**:

- **Speech Memory:** A RAM buffer with one address location per time slot per TDM highway. As each 8-bit PCM sample arrives in its time slot, it is written into the corresponding SM address.
- **Connection Memory:** A second RAM that contains the **switching instructions** — for each outgoing time slot N, the CM stores the address of the incoming time slot M from which the sample should be read. The CM is pre-loaded by the call processor when a call is set up and cleared when the call ends.
- **Output operation:** During each output time slot, the CM address for that slot is read, yielding the source SM address; the SM is then read at that address to retrieve the speech sample; the sample is placed on the outgoing TDM highway. This constitutes a **time slot interchange (TSI)** — samples are read out in a different order from how they were written in.

> **Time-Stage Operation — Example**
> ```
> Incoming TDM:  TS0[User A]  TS1[User B]  TS2[User C]  TS3[User D] ...
> Connection Memory:  OutTS0 ← InTS2   OutTS1 ← InTS0   OutTS2 ← InTS3 ...
> Outgoing TDM:  TS0[User C]  TS1[User A]  TS2[User D]  TS3[User B] ...
> ```

2.2.2 Space-Stage (S-Stage) Switching

A Space-Stage switch connects different **TDM highways** to each other. It is implemented as a **matrix of semiconductor crosspoints** (an electronic version of the analog crossbar matrix — but operating at the speed of digital electronics): an N×M matrix where N input highways can each be connected to any of M output highways during a given time slot. Different crosspoints can be active in different time slots, allowing the matrix to **reconfigure 8000 times per second** (once per PCM sampling interval) to serve different calls.

The key distinction from analog crossbar: because the space-stage matrix is clocked by the TDM system clock, a single crosspoint is only active during specific time slots — making the matrix **time-shared**. In slot N, crosspoint (i,j) may be closed to connect highway i to highway j for call A; in slot N+3, the same crosspoint — or a different one — may be active for call B.

2.3 Multi-Stage Switching: The TST and STS Architectures

A single T-stage or S-stage has limited capacity. Large digital exchanges use **multi-stage switching fabrics** combining T and S stages to achieve the required combination of high

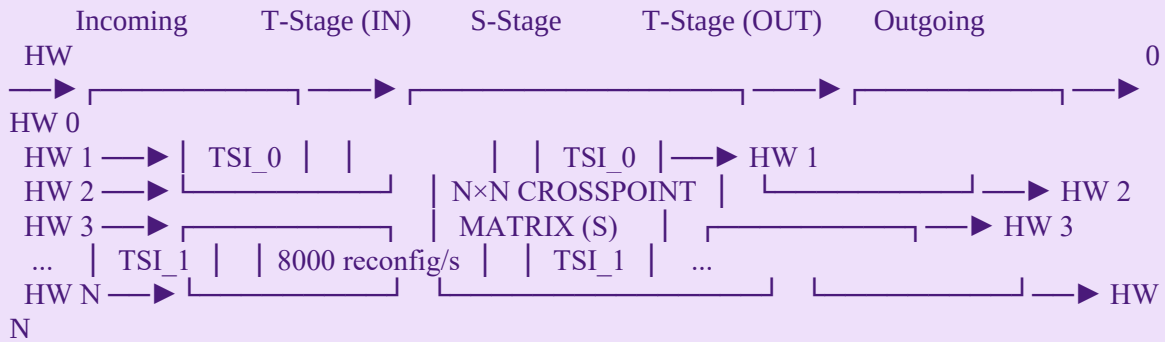
port count, low blocking probability, and manageable hardware complexity. The two canonical multi-stage architectures are:

2.3.1 TST (Time-Space-Time) Switching

TST is the most widely deployed architecture in large digital telephone exchanges (used in Alcatel E10/S12, Ericsson AXE, Nortel DMS, etc.). It consists of three cascaded stages:

1. **T-Stage (Incoming):** Performs time-slot interchange on each incoming TDM highway independently. Expands the pool of available time slots by writing incoming samples into a speech memory and reading them out in a flexible order. Converts each call from its **original time slot** to one of several possible **internal time slots** that are available in the S-Stage.
2. **S-Stage (Space):** Routes the time-multiplexed signals from any incoming TDM highway to any outgoing TDM highway during the designated internal time slot. The S-stage crosspoint matrix operates under control of a connection memory that is updated for each new call. With N inputs and N outputs and C internal time slots, the S-stage can serve N×C simultaneous connections.
3. **T-Stage (Outgoing):** Performs a second time-slot interchange on each outgoing highway to convert the signal from the internal time slot assigned by the S-stage back to the **subscriber's expected time slot** on the output TDM highway.

TST Architecture Diagram



HW = TDM Highway (e.g., 30 PCM channels per E1 link). TSI = Time-Slot Interchanger. Each TSI independently permutes time slots on its highway. The S-Stage switches between highways during each internal time slot. TST provides non-blocking switching for large N at low hardware cost.

2.3.2 STS (Space-Time-Space) Switching

STS reverses the order: S-stage first routes between input highways, a T-stage performs time-slot interchange in the internal domain, and a final S-stage routes to output highways. STS

is used in some smaller or specialized switching contexts. For very large exchanges, **five-stage or higher (e.g., TSSST) fabrics** are employed to achieve the required blocking probability targets (typically <0.001 for class 5 exchanges) while managing the physical size of each stage's crosspoint matrix.

2.4 The Call Processor and Stored-Program Control (SPC)

The switching fabric (TST/STS) provides the physical data path, but it requires a **Call Processor** — a dedicated computer running **Stored-Program Control (SPC)** software — to manage calls. The SPC:

- **Analyzes incoming signaling** (DTMF digits, SS7 ISUP messages) to determine the destination.
- **Executes routing tables** to select the appropriate output trunk or subscriber line.
- **Programs the connection memory** of the T-stages and S-stage to establish the switching path.
- **Manages call state machines** (idle, ringing, connected, releasing) for every active call simultaneously.
- **Generates billing records (CDRs)** and performance statistics for the OMC.

Modern digital exchanges are characterized by **duplicated (N+1) or fully redundant call processors** with hot standby (instantaneous switchover on failure, sub-second service disruption), achieving five-nines (99.999%) availability — equivalent to less than 5.25 minutes of downtime per year.

2.5 Softswitch (Class 5 and Class 4 Softswitches)

The **Softswitch** (also called a **Media Gateway Controller, MGC**, or **Call Agent**) is the software-based successor to the hardware TDM digital exchange. It performs the **call control logic** (SPC functions, routing, billing) in software on standard server hardware, but delegates the actual **media switching** to a **Media Gateway (MGW)** connected via a standardized protocol (MEGACO/H.248 or MGCP). This separation of call control from media switching is the **Softswitch architecture**:

- **Class 4 Softswitch:** Handles trunk-to-trunk switching between carrier networks (wholesale interconnect, tandem switching). Equivalent to a PSTN toll tandem exchange. Supports millions of simultaneous BHCA (Busy Hour Call Attempts).

- **Class 5 Softswitch:** Handles subscriber-facing services: local exchange features (call waiting, voicemail, ring groups), CLID, PSTN gateway. Equivalent to a PSTN local exchange (Class 5 CO). Interfaces directly with analog/ISDN subscriber lines via Access Media Gateways.

3. Routers in Telephony Networks

As the telephone network has progressively converged toward IP — from GPRS/UMTS packet data domains, to LTE's all-IP EPC, to SIP trunking and VoIP — the **IP router** has become a central element of telephony infrastructure. Unlike a circuit switch (which establishes a dedicated path for each call), a router forwards **individual IP packets** based on their destination address, using the packet-switched Internet Protocol.

3.1 Router Function in Telephony

In a telephony context, routers are deployed at several layers:

- **Carrier-grade aggregation routers:** In the mobile packet core (EPC S-GW/P-GW, SGi interface), massive IP routers aggregate traffic from thousands of eNBs and route it to/from the internet and IMS platform. These are **Carrier-Grade Routers (CGR)** — e.g., Cisco CRS, Juniper T-series, Nokia SR — capable of terabit/s throughput, hardware-based forwarding (ASIC/FPGA), and telco-grade redundancy (ISSU — In-Service Software Upgrade, sub-second failover via MPLS FRR).
- **IP/MPLS backbone routers:** Interconnect the operator's regional POPs (Points of Presence) and provide transport for all traffic types (voice, data, video) over a common IP/MPLS infrastructure, replacing separate TDM and ATM backbones. **MPLS Traffic Engineering (TE)** reserves bandwidth for voice and video traffic, guaranteeing QoS on the backbone.
- **Session Border Controllers (SBCs):** Specialized IP routers for VoIP, positioned at the network border to provide: NAT traversal, topology hiding, media anchoring, transcoding, interworking between SIP variants, and security (denial-of-service protection, TLS/SRTP enforcement). SBCs are functionally routers enhanced with application-layer VoIP intelligence.

- **Enterprise IP PBX routers/gateways:** At the enterprise level, IP-PBX systems (e.g., Cisco CUCM, Avaya CM, Asterisk) use SIP trunks over IP routers to connect to carrier networks, replacing legacy T1/E1 PRI circuits.

3.2 VoIP Packet Forwarding vs. Circuit Switching

The fundamental operational difference between a circuit switch and an IP router for telephony traffic:

Property	Circuit Switch (TDM)	IP Router (VoIP)
Path establishment	Dedicated end-to-end circuit set up before first byte transmitted; resources reserved for entire call duration	No path setup; each packet routed independently based on destination IP address; no prior reservation
Resource allocation	Fixed 64 kbit/s (G.711) per call; channel occupied even during silence	Statistical multiplexing; bandwidth used only when packets present; VAD/CNG reduces bandwidth ~50%
Delay	Constant, predictable, very low (< 1 ms switching delay); deterministic	Variable (jitter): depends on queue depth, routing, congestion; mitigated by jitter buffer at endpoint
Packet loss	No packet loss in circuit switching (physical continuity); errors cause signal degradation only	Packets may be dropped under congestion; loss > 1–2% degrades voice quality; mitigated by QoS priority queuing
Addressing	Physical/logical circuit ID (time slot number, trunk ID)	IP destination address (32-bit IPv4 or 128-bit IPv6) + UDP/RTP port number
Signaling	SS7 (ISUP, MAP, INAP) on separate signaling network (CCS7)	SIP (Session Initiation Protocol) or H.323 over IP; media described by SDP
Efficiency under load	Hard capacity limit (N circuits max); blocking when all circuits occupied	Soft capacity; all calls share bandwidth; quality degrades under congestion without QoS enforcement

3.3 QoS Mechanisms for VoIP over IP Networks

Voice over IP is **highly sensitive to latency, jitter, and packet loss**: ITU-T G.114 recommends one-way delay < 150 ms for satisfactory quality; jitter > 50 ms is perceptible; packet loss > 2–3% causes audible gaps and clipping. Without Quality of Service (QoS) mechanisms, voice packets share bandwidth with best-effort data traffic (file downloads, web browsing) and are subject to the same queuing delays and drops.

QoS in IP telephony is implemented at multiple layers:

3.3.1 DSCP Marking (DiffServ)

The **Differentiated Services Code Point (DSCP)** is a 6-bit field in the IP header (within the ToS/Traffic Class byte). Voice RTP packets are marked **EF (Expedited Forwarding, DSCP 46 = 101110 binary)**, indicating that these packets should receive the lowest latency, lowest jitter treatment at every router. Routers are configured to process EF-marked packets before all other traffic classes. SIP signaling packets are typically marked **AF31 or CS3** (higher priority than best-effort but below EF voice media).

3.3.2 Priority Queuing and Scheduling

Each router interface maintains multiple output queues. **LLQ (Low Latency Queuing)** — also called Priority Queuing (PQ) — allocates a dedicated queue for EF-marked voice packets that is **always served before other queues** (strict priority). This guarantees bounded delay for voice packets regardless of data traffic load. A **bandwidth police mechanism** (traffic shaper on the LLQ) prevents voice traffic from starving data queues entirely by capping the EF queue's bandwidth share (e.g., maximum 30% of link bandwidth for voice).

3.3.3 RSVP (Resource Reservation Protocol)

For networks requiring guaranteed end-to-end QoS (not merely best-effort prioritization), **RSVP** (defined in RFC 2205) allows a VoIP application to explicitly **reserve bandwidth** on each router along the path between sender and receiver before the call begins (call admission control at the network level). RSVP PATH and RESV messages traverse the route, and each router either grants the reservation (if bandwidth is available) or rejects it (triggering a **call admission rejection** — the VoIP equivalent of a busy signal). RSVP is used in high-quality enterprise and carrier VoIP deployments but is rarely deployed on the public internet.

3.3.4 Jitter Buffer

Although not a router mechanism, the **jitter buffer** at the VoIP endpoint (IP phone or gateway) is essential for mitigating network jitter. Received RTP packets are held in a buffer and played out at a fixed rate, smoothing out the variable arrival times. An **adaptive jitter buffer** dynamically adjusts its depth: small buffer (20–60 ms) for low-jitter paths (lower added delay); large buffer (100–200 ms) for high-jitter paths (higher added delay but fewer buffer underflows/overflows). The jitter buffer depth adds directly to the one-way voice delay and must be managed within the G.114 150 ms budget.

> ⚠ **VoIP Quality: The Impairment Triangle**
>
> **Delay:** One-way delay > 150 ms (G.114) causes conversational awkwardness. > 400 ms is unacceptable. Sources: algorithmic delay (codec lookahead), packetization delay (20 ms typical), jitter buffer, network propagation, queuing.
>
> **Jitter:** Variable packet delay. > 50 ms perceptible as voice distortion and gaps. Mitigated by adaptive jitter buffer at cost of added delay.
>
> **Packet Loss:** > 1% causes audible gaps; > 5% severe degradation. Concealed by PLC (Packet Loss Concealment) in the codec (e.g., G.729 Annex I, G.711 Appendix I). Burst loss is more damaging than random loss at the same average rate. Measured by MOS (Mean Opinion Score, 1–5) using PESQ (ITU-T P.862) or E-Model (G.107).

4. Gateways (Passerelles)

Gateways are the **critical interconnection devices** that enable communication between networks built on fundamentally different technologies. In the telephony context, the most important distinction is between the **circuit-switched PSTN** (TDM trunks, SS7 signaling, G.711 PCM voice) and the **IP/VoIP domain** (RTP packets, SIP signaling, compressed codecs). Gateways bridge this divide at both the **media plane** (voice samples) and the **signaling plane** (call control messages).

4.1 Media Gateway (MGW)

4.1.1 Definition and Role

A **Media Gateway (MGW)** is the network element responsible for **converting voice media between circuit-switched TDM formats and IP/RTP packet formats**. It is the physical/logical point where a PCM time slot from the PSTN is converted into an IP packet stream carrying the same conversation, and vice versa. The MGW has no call control intelligence of its own — it is controlled by a **Media Gateway Controller (MGC)**, also called a **Softswitch**, via the **H.248/MEGACO** protocol (standardized by IETF RFC 3525 and ITU-T H.248.1).

4.1.2 MGW Functions

- **TDM/RTP Conversion:** Terminates TDM circuits (E1/T1 ports, typically carrying 30/24 PCM channels respectively) on one side, and IP interfaces (GbE, 10GbE) on the other. For each active call, it reads 8-bit PCM samples from the incoming TDM time slot, packetizes them into **RTP (Real-time Transport Protocol)** packets (typically 20 ms of audio per packet = 160 bytes G.711 payload), and transmits them as UDP/IP datagrams to the destination VoIP endpoint or gateway.
- **Codec Transcoding:** Converts between different voice encoding formats: G.711 μ-law (North America) ↔ G.711 A-law (Europe/international), G.711 ↔ G.729A (8 kbit/s, compressed), G.711 ↔ G.723.1 (5.3/6.3 kbit/s), G.711 ↔ AMR (mobile networks). Transcoding requires DSP (Digital Signal Processor) hardware and introduces additional algorithmic delay (typically 5–20 ms per codec stage).
- **Echo cancellation:** Hybrid circuits at the 2-wire/4-wire conversion point of the PSTN create acoustic echo that is particularly problematic at the high delays of VoIP paths. The MGW implements **echo cancellers** per ITU-T G.168, with tail lengths up to 128 ms, using adaptive LMS filters to subtract the echo from the near-end talker's signal.
- **Tone detection and generation:** Detects in-band DTMF tones from the PSTN side (RFC 2833 / RFC 4733 out-of-band event reporting to the IP side), and generates PSTN tones (dial tone, busy, ring-back) toward the TDM side for incomplete calls.
- **Fax/modem interworking:** Standard voice codecs destroy modem and fax signals through lossy compression. The MGW detects T.38 fax relay and V.151/V.152 modem relay signals and re-encapsulates them using appropriate IP protocols (T.38 for fax, relay for modem).

4.1.3 Access vs. Trunking Media Gateways

- **Trunking Media Gateway:** Converts between carrier-grade TDM trunks (E1/T1 interfaces carrying ISUP calls) and IP/SIP trunks. Located at the border between the PSTN and the VoIP core (e.g., between an incumbent operator's SS7 network and a VoIP interconnect platform). May handle thousands of simultaneous calls (several hundred E1 ports, up to 7,500+ simultaneous calls per chassis).
- **Access (Residential) Media Gateway:** Connects individual analog subscriber lines (POTS, FXS ports) or small ISDN BRI/PRI interfaces to the IP network. Used in DSLAM-adjacent equipment, cable modem gateways (EMTA — Embedded

Multimedia Terminal Adapter, DOCSIS-based), and enterprise IADs (Integrated Access Devices) to migrate legacy analog phones to VoIP without replacing the handset.

4.2 Signaling Gateway (SGW)

4.2.1 Definition and Role

A **Signaling Gateway (SGW)** is the network element that performs **protocol translation between SS7 (used in the PSTN/mobile CS domain) and SIP (used in the VoIP/IMS domain)** at the signaling (control) plane level. While the MGW handles media, the SGW handles the call control messages that set up, manage, and release calls across the PSTN-VoIP boundary.

4.2.2 SS7 to SIP Translation

The SS7 signaling network uses a specialized protocol stack (MTP1/2/3 + SCCP + TCAP + ISUP/MAP, as studied in Chapter 3) transported over dedicated 64 kbit/s signaling links between SS7 Signal Transfer Points (STPs). The VoIP/IMS domain uses SIP over IP/UDP or TCP. The SGW must:

4. **Transport adaptation:** Terminate SS7 MTP links (E1 D-channels or IP-encapsulated SS7 via **SIGTRAN** — e.g., M3UA over SCTP/IP as defined in IETF RFC 4666) and bridge them to the IP-based SIP signaling network.
5. **ISUP ↔ SIP message mapping:** Map SS7 ISUP call control messages to SIP equivalents and vice versa:
    - **ISUP IAM** (Initial Address Message) → **SIP INVITE** (with called/calling number in SIP headers, SDP offer for media)
    - **ISUP ACM** (Address Complete Message) → **SIP 180 Ringing**
    - **ISUP ANM** (Answer Message) → **SIP 200 OK**
    - **ISUP REL** (Release) → **SIP BYE**
    - **ISUP RLC** (Release Complete) → **SIP 200 OK** response to BYE
6. **Parameter translation:** Map ISUP parameters (Called Party Number in E.164, Calling Party Number, Bearer Capability, Nature of Address Indicator) to corresponding SIP headers and SDP parameters (To:/From:/Contact: URIs, Supported: codec list).

7. **Cause code mapping:** Map ISUP Release Cause values (e.g., Cause 16 = Normal clearing, Cause 34 = No circuit available) to corresponding SIP response codes (200, 486 Busy Here, 503 Service Unavailable, 603 Decline).

SIGTRAN: SS7 over IP

SIGTRAN is the IETF working group that standardized protocols for transporting SS7 signaling over IP networks. Key protocols:

SCTP (RFC 4960): Stream Control Transmission Protocol — a transport layer protocol designed specifically for signaling transport. Provides: multi-homing (a single association can use multiple IP addresses for redundancy), multi-streaming (independent ordered delivery of multiple message streams), message-oriented delivery (preserving SS7 MSU boundaries), and built-in heartbeat for path monitoring.

M3UA (RFC 4666): MTP3 User Adaptation layer — transports MTP3 user messages (ISUP, SCCP) over SCTP/IP to a Signaling Gateway that connects to the SS7 network.

M2PA / M2UA: MTP2 Peer Adaptation / User Adaptation — transport individual SS7 links over SCTP/IP at lower granularity (MTP2 level), used for interconnecting SS7 STPs over IP.

4.3 Other Gateway Types

- **H.323 ↔ SIP Gateway:** Translates between H.323 (ITU-T standard, widely deployed in early VoIP enterprise systems and video conferencing) and SIP. The two protocols have different session description mechanisms (H.245 capability exchange vs. SDP offer/answer) and addressing (H.323 gatekeeper vs. SIP proxy/registrar). Interworking gateways map H.323 call flows to SIP flows and H.245 media capability exchanges to SDP.
- **ISDN/QSIG Gateway:** Converts between ISDN Q.931 signaling (used for PBX-to-PBX QSIG or ISDN BRI/PRI subscriber access) and SIP. Used in enterprise telephony migration, connecting legacy ISDN PBXs to modern IP telephony platforms.
- **IMS Application Server (AS) / BGCF:** In the LTE/IMS architecture, the **BGCF (Breakout Gateway Control Function)** determines where a call should break out to the PSTN, and the **MGCF (Media Gateway Control Function)** controls the MGW for PSTN interworking, together acting as an integrated SGW+MGC.

5. Interconnection Interfaces

Interconnection between telephone exchanges, gateways, and routers requires precisely specified **physical interfaces** (electrical/optical connectors, line codes, signal levels) and **protocol interfaces** (frame structures, signaling protocols). These interfaces are rigorously standardized by ITU-T, IETF, ETSI, and ANSI to ensure interoperability between equipment from different manufacturers.

5.1 E1 / T1 Primary Rate Interfaces (PRI)

5.1.1 E1 Interface (ITU-T G.703 / G.704 — Europe, Africa, Asia)

The **E1 interface** is the standard digital trunk interface in European, African, and Asian telephone networks. It carries **30 voice channels + 2 overhead channels** at a line rate of **2.048 Mbit/s** using **HDB3 line coding** (High-Density Bipolar of order 3: suppresses long runs of zeros while maintaining AMI bipolar rule for clock recovery) over a balanced 120 Ω twisted-pair cable or 75 Ω coaxial cable.

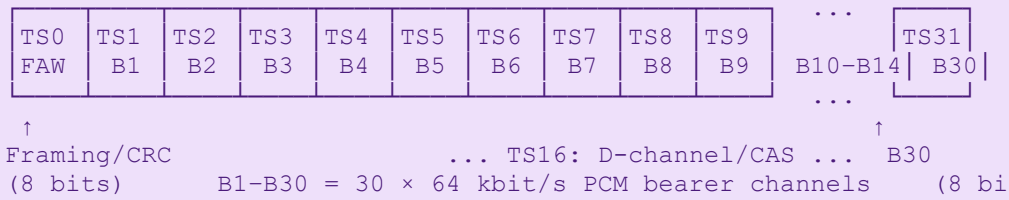
5.1.2 E1 Frame Structure (G.704)

The E1 frame structure consists of:

- **Frame period: 125 µs** (= 1/8000 Hz, one PCM sampling interval per G.711).
- **32 time slots per frame** (TS0 through TS31), each 8 bits wide × 8000 frames/s = 64 kbit/s per time slot.
- **TS0: Framing and Synchronization**. Alternates between a **frame alignment word (FAW: 0011011)** in even frames and a **non-frame alignment word** + CRC (in CRC-4 multiframe mode, G.704 Section 2.3) in odd frames. The remote end uses FAW to achieve frame synchronization (byte alignment).
- **TS16: Signaling (CAS) or D-channel (CCS)**. In **Channel Associated Signaling (CAS / R2)** mode: carries 4-bit ABCD signaling nibbles for each of the 30 speech channels (2 channels per frame, across a 16-frame multiframe). In **Common Channel Signaling (CCS / ISUP / QSIG / DSS1 PRI)** mode: TS16 carries a 64 kbit/s D-channel (Layer 2: LAPD; Layer 3: Q.931 for DSS1/ISDN PRI, or SS7 MTP2 for SS7-over-E1 links).

- **TS1–TS15, TS17–TS31: 30 B-channels**. Each carries one 64 kbit/s PCM voice channel (or data channel). In ISDN PRI terminology: **30B+D** (30 Bearer channels + 1 D-channel signaling channel).

E1 Frame Structure Diagram



Total: 32 × 8 bits = 256 bits/frame × 8000 frames/s = 2.048 Mbit/s. Frame duration = 125 µsec. CRC-4 multiframe = 16 frames = 2 ms.

5.1.3 T1 Interface (ANSI T1.403 — North America, Japan)

The **T1 interface** is the North American / Japanese equivalent of E1:

- **Line rate: 1.544 Mbit/s** (24 DS0 channels × 64 kbit/s + 8 kbit/s framing overhead).
- **Frame structure: 24 time slots** (TS1–TS24, each 8 bits), plus **1 framing bit** per frame = 193 bits/frame × 8000 frames/s = 1.544 Mbit/s.
- **Framing:** D4 (Superframe, 12-frame grouping) or **ESF (Extended SuperFrame, 24-frame grouping)**. ESF uses the framing bit for: 6 bits of framing pattern, 6 bits CRC-6, and 12 bits data link channel (DL: 4 kbit/s for OAM).
- **Signaling:** CAS ("Robbed-bit signaling": LS bit of every 6th frame stolen for ABCD signaling) or CCS (23B+D — 23 bearer channels + 1 64 kbit/s D-channel for ISDN PRI in North America).
- **Line coding: AMI** (Alternate Mark Inversion) or **B8ZS** (Bipolar 8-Zero Substitution, for ESF to maintain ones-density).

5.1.4 Signaling Protocols over E1/T1: ISUP and DSS1

- **ISUP (ISDN User Part):** The SS7 application protocol for **trunk-level call control** between telephone exchanges (inter-exchange signaling). Rides on top of MTP3 (or M3UA/SCTP/IP for SIGTRAN). Defines messages for circuit-switched call setup: IAM (Initial Address Message), SAM (Subsequent Address Message), ACM, ANM, REL, RLC, CPG (Call Progress), etc. Each ISUP call is associated with a **Circuit**

Identification Code (CIC) that uniquely identifies the E1/T1 bearer circuit (time slot) carrying the voice.

- **DSS1 / Q.931 (Digital Subscriber Signaling System No. 1):** The ISDN **subscriber-access** signaling protocol, used between a subscriber terminal (ISDN phone, PBX) and the local exchange over the D-channel of an ISDN BRI (2B+D, 192 kbit/s) or PRI (30B+D / 23B+D). Q.931 messages (SETUP, CALL PROCEEDING, ALERTING, CONNECT, RELEASE, DISCONNECT) are functionally similar to SIP but use binary encoding rather than text.
- **CAS / R2 Signaling:** The oldest and simplest form of inter-exchange signaling still in use in many developing-country telephone networks. Line state (idle, seized, busy) is conveyed by the ABCD bits in TS16 (E1) or robbed bits (T1). R2 (ITU-T Q.421) uses in-band Multi-Frequency (MF) tones for digit transfer (similar to DTMF but separate forward and backward tone groups for call control dialogues). No SS7 required.

5.2 SIP Trunks (VoIP Trunks over IP)

A **SIP trunk** is the VoIP equivalent of a bundle of E1/T1 circuits: a logical connection between two VoIP platforms (IP-PBX to carrier, or carrier-to-carrier) that carries multiple simultaneous voice calls as SIP sessions over IP/UDP or TCP. SIP trunks have progressively replaced E1/T1 PRI circuits in enterprise and carrier environments due to several advantages:

- **No physical channel limit:** The number of simultaneous calls on a SIP trunk is not fixed at installation but determined by the licensed capacity of the terminating platform and the available IP bandwidth. Scaling up requires a software configuration change, not hardware installation.
- **Cost efficiency:** Eliminates per-circuit hardware (E1 interface cards, cross-connect panels) and enables **Local Number Portability (LNP)** and geographic flexibility (a "trunk" can terminate on a server in any city).
- **Codec flexibility:** SIP trunks support multiple codecs negotiated per call via SDP: G.711 (highest quality, 64 kbit/s), G.729A (8 kbit/s, transparent to most users), G.722 (wideband/HD voice, 16 kHz bandwidth — noticeably improved quality), Opus (3–510 kbit/s, internet-native, 3GPP EVS for 5G).
- **Security requirements:** SIP trunks over public IP require: **TLS** (Transport Layer Security) for SIP signaling encryption, **SRTP** (Secure Real-time Transport Protocol) for

media encryption, and an **SBC** (Session Border Controller) at the network border for NAT traversal, topology hiding, and DoS protection.

5.3 Gigabit Ethernet and Optical Fiber Transport

5.3.1 IP Trunk Transport over Ethernet / Fiber

In modern IP-based telephony networks, **Gigabit Ethernet (GbE, IEEE 802.3z)** and **10 Gigabit Ethernet (10GbE, IEEE 802.3ae)** or higher (25GbE, 100GbE, 400GbE) over single-mode optical fiber (LC/SC connector, 1310/1550 nm) have replaced E1 bundles and ATM links as the primary transport medium for interconnecting:

- **eNBs / gNBs to EPC:** The S1 interface in LTE and NG interface in 5G are IP-based over GbE or 10GbE fiber. A single 10GbE fiber link replaces hundreds of E1 circuits in cellular backhaul.
- **SBC / SIP trunk connections:** SIP trunks are delivered to enterprise premises or carrier interconnection points over GbE Ethernet handoff (typically on dedicated VLANs, 802.1Q tagged, with DSCP marking honoured across the interconnect).
- **DWDM backbone transport:** Long-haul and metropolitan optical transmission (studied in Chapter 2) carries all telephony traffic (SS7 signaling, VoIP, mobile backhaul, PSTN trunks encapsulated in SDH/SONET or OTN) over wavelength-division-multiplexed fiber.

5.3.2 SDH / OTN Framing for Telephony Trunks

Traditional E1/T1 circuits are aggregated into **PDH (Plesiochronous Digital Hierarchy)** frames (E2/E3/E4: 8.448, 34.368, 139.264 Mbit/s) and transported within **SDH (Synchronous Digital Hierarchy, ITU-T G.707/G.803)** containers: VC-12 carries one E1; VC-4 carries 63 E1s; STM-1 (155.52 Mbit/s) carries one VC-4; STM-16 (2.5 Gbit/s) carries 16. SDH provides precise synchronization (stratum clocking hierarchy), protection switching (< 50 ms restoration via APS — Automatic Protection Switching), and rich OAM capabilities.

Modern networks use **OTN (Optical Transport Network, ITU-T G.709)** — which encapsulates Ethernet, SDH, and other client signals into standardized optical container frames (OTU1/2/3/4) with strong FEC (Forward Error Correction, up to 15–25% overhead) for ultra-long-haul transmission, and nested tandem connections for multi-operator service monitoring.

5.4 Comparative Table: Interconnection Equipment and Interfaces

Equipment / Interface	Technology Domain	Data Rate / Capacity	Key Protocol(s)	Primary Interconnection Function
Analog crossbar switch	PSTN (analog CS)	N×M circuits, 3.1 kHz voice	Pulse/DTMF dialing	Subscriber line concentration and circuit path establishment (legacy)
Digital switch (TST/STS)	PSTN (digital CS)	10K–500K+ circuits, 64 kbit/s PCM	SS7/ISUP, DSS1, CAS	Digital time-space switching; call routing; concentration/expansion
Softswitch (MGC/Class 4/5)	VoIP / PSTN boundary	Millions BHCA; millions of sessions	SIP, H.248/MEGACO, SS7	Software-based call control; replaces digital TDM exchange hardware
Carrier-Grade Router (CGR)	IP / MPLS core	1–100+ Tbit/s throughput	IP, MPLS, BGP, OSPF, LDP	IP packet forwarding; MPLS TE for VoIP QoS; EPC user-plane transport
Session Border Controller (SBC)	VoIP border / IP-IP	Tens of thousands SIP sessions	SIP, RTP, SRTP, TLS	NAT traversal; topology hiding; transcoding; DoS protection; SIP normalization
Trunking Media Gateway (MGW)	PSTN ↔ VoIP	Hundreds of E1s / thousands of calls	H.248/MEGACO, RTP, G.711/G.729	TDM ↔ IP media conversion; transcoding; echo cancellation; DTMF relay
Access Media Gateway (IAD/EMTA)	Analog POTS ↔ VoIP	2–24 FXS ports; 1–2 FXO ports	SIP/MGCP, RTP, G.711/G.729	Analog subscriber loop → IP; enables VoIP on legacy handsets;

Equipment / Interface	Technology Domain	Data Rate / Capacity	Key Protocol(s)	Primary Interconnection Function
				DSLAM or cable MSO edge
Signaling Gateway (SGW)	SS7 ↔ SIP / SIGTRAN	Thousands of SS7 MSUs/s; thousands SIP dialogs/s	SS7/ISUP, SIGTRAN (M3UA/SCTP), SIP	ISUP ↔ SIP signaling protocol translation; SS7 point code ↔ SIP URI mapping
E1 / 30B+D PRI	PSTN digital trunk	2.048 Mbit/s; 30 × 64 kbit/s	ISUP, DSS1/Q.931, CAS/R2	Physical bearer for 30 PCM circuits + D-channel signaling between exchanges or PBX-CO
T1 / 23B+D PRI	PSTN digital trunk (NA)	1.544 Mbit/s; 24 × 64 kbit/s	ISUP, Q.931, B8ZS, ESF	North American equivalent of E1; 23 B-channels + 1 D-channel (ISDN PRI)
SIP Trunk (IP trunk)	VoIP (IP-IP)	Unlimited (bandwidth-limited); G.711/G.729/Opus	SIP, SDP, RTP/SRTP, TLS	Virtual trunk; replaces E1 PRI; flexible capacity; requires SBC at border
GbE / 10GbE over SMF	IP/Ethernet transport	1/10/25/100/400 Gbit/s	IEEE 802.3, VLAN 802.1Q, DSCP	Physical transport for SIP trunks, S1/NG backhaul, OAM; replaces E1 bundles
SDH / OTN (STM-n / OTUk)	Optical backbone transport	155 Mbit/s (STM-1) to 100+ Gbit/s (OTU4)	G.707 / G.709, APS, FEC	Multiplexes E1/Ethernet/IP over fiber; < 50 ms protection switching; submarine cables

6. Chapter Summary: Key Concepts to Remember

Concept / Term	Essential Definition / Key Point
Interconnection Purpose	Enables communication across heterogeneous networks (analog CS, digital CS, VoIP/IP) by converting physical media, signaling protocols, and voice encoding formats at their boundaries.
Time-Stage Switching (TSI)	Routes PCM samples between time slots on the same TDM highway using Speech Memory + Connection Memory. CM stores switching instructions per output slot; written by call processor at call setup.
Space-Stage Switching	Electronic crosspoint matrix connecting different TDM highways. Matrix reconfigures 8000 times/second (once per PCM frame). Different crosspoints active in different time slots for different calls.
TST Architecture	Time-Space-Time: three-stage digital switch fabric. T-stage in: flexibly re-orders time slots to find available internal slots. S-stage: routes between highways. T-stage out: maps to subscriber's expected outgoing slot.
Softswitch / MGC	Software-based call processor on standard server hardware. Controls Media Gateways via H.248/MEGACO. Separates call control (Softswitch) from media switching (MGW). Class 4: inter-carrier tandem. Class 5: subscriber services.
VoIP vs. Circuit Switch	Circuit switch: dedicated path, fixed 64 kbit/s, deterministic delay, hard capacity limit. IP router: statistical multiplexing, variable delay/jitter, packet loss risk, soft capacity — requires QoS to match circuit switch voice quality.
DSCP / EF Marking	VoIP RTP packets marked DSCP EF (46) for Expedited Forwarding. SIP signaling: AF31 or CS3. Routers serve EF queue first (strict priority/LLQ), guaranteeing bounded delay for voice.
VoIP Quality Triangle	Three impairments: Delay (< 150 ms one-way per G.114), Jitter (mitigated by adaptive jitter buffer, adds delay), Packet Loss (< 1–2%, concealed by PLC). Measured by MOS (1–5) via PESQ or E-Model.
Media Gateway (MGW)	Converts TDM voice (E1/T1 PCM slots) to/from IP/RTP packets. Also: transcoding (G.711 ↔ G.729/AMR), echo cancellation (G.168), DTMF relay (RFC 4733), fax relay (T.38). Controlled by MGC via H.248.
Signaling Gateway (SGW)	Translates SS7/ISUP signaling to SIP and vice versa. Maps: IAM→INVITE, ACM→1xx, ANM→200 OK, REL→BYE. Uses SIGTRAN (M3UA/SCTP/IP) for SS7-over-IP transport.
SIGTRAN (M3UA / SCTP)	IETF protocol for transporting SS7 over IP. SCTP: multi-homed, message-oriented transport with heartbeat. M3UA: maps MTP3 primitives to SCTP associations. Enables SS7 inter-working without dedicated TDM signaling links.

Concept / Term	Essential Definition / Key Point
E1 Interface (G.703/G.704)	2.048 Mbit/s, 32 × 64 kbit/s time slots, 125 µs frame, HDB3 line code. TS0: FAW+CRC4. TS16: D-channel (ISUP/Q.931) or CAS signaling. TS1–15/TS17–31: 30 B-channels (30B+D ISDN PRI).
T1 Interface (ANSI T1.403)	1.544 Mbit/s, 24 × 64 kbit/s + framing bit = 193 bits/frame. ESF (24-frame SF) with CRC-6 + 4 kbit/s DL. 23B+D (ISDN PRI NA). B8ZS line code for ones-density.
ISUP (SS7 trunk signaling)	ISDN User Part: inter-exchange call control over SS7. Key messages: IAM (call request + called/calling number), ACM (ringing), ANM (answer), REL (hang-up), RLC (circuit released). Each call identified by CIC (circuit ID).
DSS1 / Q.931 (ISDN PRI)	Subscriber access signaling over D-channel (Layer 2: LAPD; Layer 3: Q.931). Messages: SETUP, CALL PROC, ALERT, CONNECT, DISCONNECT, RELEASE. Used for PBX-to-CO and ISDN enterprise telephony.
SIP Trunk	VoIP replacement for E1/T1 PRI. Logical IP connection; unlimited simultaneous calls (bandwidth-limited). Requires SBC (NAT, security), TLS (signaling), SRTP (media). Supports HD voice (G.722, EVS).
Session Border Controller (SBC)	Application-aware VoIP border router: NAT traversal, topology hiding, SIP normalization, transcoding, TLS/SRTP enforcement, DoS protection. Mandatory at all SIP trunk boundaries.
SDH / OTN Transport	SDH (G.707): synchronous multiplexing of E1s and Ethernet over fiber; < 50 ms APS protection. OTN (G.709): encapsulates Ethernet/SDH with strong FEC for submarine/long-haul; nested performance monitoring.

Module Conclusion: The Complete Telephony Ecosystem

This final chapter completes the Telephony module's technical arc. The five chapters together tell a coherent engineering story: Chapter 1 established the foundational concept of circuit switching and the analog PSTN; Chapter 2 examined the physical transmission media that carry all signals; Chapter 3 explored GSM 2G digital cellular architecture and protocols; Chapter 4 traced the evolution to UMTS, LTE, and 5G broadband mobile generations; and Chapter 5 has now addressed the interconnection equipment and interfaces that bind all these networks together into a seamless global telecommunications system. The progressive convergence toward IP — from TDM circuit switching to VoIP over packet networks, from SS7 to SIP, from hardware exchanges to cloud-native Softswitches — is the defining engineering trend of the past three decades,

and the equipment and interfaces studied in this chapter are the enabling mechanisms of that transformation.

GENERAL CONCLUSION

I. A Journey Across 150 Years of Telecommunications Engineering

This module has taken us on a journey across one hundred and fifty years of telecommunications engineering, from the first crackling voice transmission over Bell's primitive telephone to the terabit-capacity 5G networks that are currently being deployed around the world. That this journey could be encompassed within five chapters is a testament not to the simplicity of the subject, but to the coherence of the underlying engineering principles: the same fundamental requirements — to establish a reliable communication path, to transmit voice signals with acceptable quality, to share scarce physical resources efficiently, to scale the system to serve millions of simultaneous users — have driven every generation of telephony innovation from the electromechanical relay to the silicon ASIC to the cloud-native microservice.

II. Key Engineering Lessons from Each Chapter

Chapter 1: The Enduring Logic of Circuit Switching

The study of analog switching telephony revealed a foundational truth of communications engineering: **the topology of the network must match the nature of the traffic**. Voice communication is inherently **symmetric, real-time, and continuous**; circuit switching — the dedication of a fixed-bandwidth path for the entire duration of a call — was the natural and correct solution for the technology of the 19th and 20th centuries. The Strowger step-by-step switch was a brilliant mechanical implementation of a directory lookup; the crossbar switch extended this to a parallel, non-blocking matrix; the TST digital switch fabric abstracted the concept into the time domain. The three phases of a circuit-switched call (setup, transfer, teardown) and the hierarchical PSTN exchange structure (Class 1 international through Class 5 local) remain the conceptual template against which all subsequent switching paradigms are compared.

Chapter 2: The Physics of the Channel Determines the System

Transmission media are not neutral conduits: their physical properties **fundamentally constrain** the services that can be delivered over them. The frequency-dependent attenuation of

copper twisted pair — growing as √f — explains why DSL rates fall sharply with distance and why Fiber-to-the-Home was an economic and technical necessity. The 240 ms one-way propagation delay of geostationary satellites — a direct consequence of orbital mechanics — makes GEO satellite circuits permanently unsuitable for real-time voice without echo cancellation. The 0.2 dB/km loss minimum of single-mode silica fiber at 1550 nm — a property of the Rayleigh scattering floor and infrared absorption tail of SiO_2 — is what makes transoceanic fiber cables feasible. The cellular principle, with its D/R = √(3N) reuse distance formula, is the elegant geometric solution to the problem of fitting more telephone calls into a finite radio spectrum by exploiting the physical attenuation of radio waves with distance. Every major system design decision in telephony has been shaped, at its root, by the physics of the transmission medium.

Chapter 3: Architecture Follows Function

The GSM architecture is a masterpiece of modular engineering design: each of its three subsystems (BSS, NSS, OMS) addresses a distinct and well-bounded set of functions; each interface between subsystems (Um, Abis, A, B, C, D, E, F, G, H) is precisely standardized to allow multi-vendor interoperability; and each network element (BTS, BSC, MSC, HLR, VLR, AuC, EIR) has a clearly defined role in the overall system. The decision to separate **radio resource management (BSC)** from **mobility management (VLR)** from **call control (MSC)** from **subscriber data management (HLR)** — and to house authentication credentials in the physically secure AuC and SIM — reflects deep architectural wisdom that has survived three decades of network evolution. The GSM protocol stack, with its clean separation into Layer 1 (GMSK physical), Layer 2 (LAPDm), and Layer 3 (RR/MM/CM), demonstrated that the OSI model is a practical engineering tool, not merely a theoretical abstraction. The KPIs of Chapter 3 — RSSI, RxQual, CDR, CSSR — are the operational instruments through which abstract architecture becomes measurable reality.

Chapter 4: Innovation is Driven by the Gap Between Need and Capability

The progression from GSM to UMTS to HSPA to LTE to 5G NR is a textbook illustration of how technology evolves when the gap between what users need and what the current technology can deliver becomes commercially and socially unacceptable. GSM's 9.6 kbit/s data rate was adequate for SMS in 1992; it was hopelessly insufficient for the mobile web of 2001. W-CDMA's 384 kbit/s sufficed for early mobile internet; it could not support the video streaming demand unleashed by 3G-era smartphones. LTE's 150 Mbit/s peak rate served the first generation

of HD mobile video consumers; it struggles with the simultaneous connectivity demands of stadium crowds and the millisecond latency requirements of industrial automation. Each technological leap — W-CDMA spreading (processing gain against narrowband interference), HARQ soft combining (information-theoretically optimal retransmission), OFDMA (ISI immunity through long symbol duration), massive MIMO (spatial multiplexing gain), network slicing (logical isolation of heterogeneous services) — was a targeted engineering solution to a specific physical or economic constraint.

Chapter 5: Integration is the Hardest Problem

The existence of specialized interconnection equipment — TST digital switches, softswitches, media gateways, signaling gateways, session border controllers, SIP trunks, SIGTRAN — is testimony to the hardest challenge in telecommunications engineering: **making different systems talk to each other**. Building a new network from scratch is an engineering problem with a well-defined solution space. Interconnecting a new IP network with a legacy SS7-based PSTN, while preserving every existing subscriber's ability to make and receive calls, while not disrupting the billing systems that depend on CDR formats from 1985, while maintaining the five-nines availability that regulators mandate — this is an integration challenge of a different order of complexity. The SIGTRAN protocol family, the H.248/MEGACO architecture, and the SBC concept are remarkable examples of engineering pragmatism: solutions that were not theoretically optimal but were practically deployable, interoperable, and economically viable at the required scale.

III. The Convergence Horizon: Telephony and Computing Become One

The most significant trend emerging from this module's five chapters is the progressive and apparently irreversible **convergence of telephony and computing**. In 1990, a telephone switch was a specialized piece of hardware with proprietary software, physically separate from the data network. In 2025, a 5G core network function is a software container running on a commercial off-the-shelf (COTS) server in a data centre, exposed as a microservice via a REST API over HTTP/2, deployed and scaled using Kubernetes, monitored using Prometheus, and interconnected via gRPC — indistinguishable from a web application in its implementation technology.

This convergence has profound implications for the telecommunications engineering profession. The boundaries between network engineering, software engineering, cloud infrastructure, and cybersecurity are dissolving. A modern 5G network engineer must understand not only radio propagation and protocol stacks, but also containerization, DevOps, zero-trust security architectures, and machine learning (for AI-native 6G air interfaces). The foundational concepts mastered in this module — switching, multiplexing, signaling, quality measurement, protocol architecture — provide the essential substrate for engaging with these advanced topics, because the problems they address — establishing connections, sharing resources, ensuring quality, authenticating identities, and maintaining service continuity — are **permanent** problems of telecommunications, independent of the implementation technology.

IV. Looking Ahead: From 5G to 6G and the Intelligent Network

As this module concludes, the telecommunications research community is already engaged in the conceptual and early standardization work for **6G (IMT-2030)**, with commercial deployments expected around 2030. While 6G's technical specifications are still being debated, its motivating vision is clear: a communication system in which the **radio channel itself becomes programmable** (through Reconfigurable Intelligent Surfaces), in which the **air interface is learned rather than hand-designed** (through AI-native physical layer algorithms), in which **terahertz frequencies** provide extreme bandwidth for short-range applications, and in which **the network simultaneously senses and communicates** (Integrated Sensing and Communication, ISAC). Behind all these innovations lies a single engineering imperative that connects 6G to Bell's first telephone: the imperative to connect every human being and every intelligent machine to every other, with the highest possible fidelity, the lowest possible latency, and the most efficient possible use of the physical and electromagnetic resources of our world.

The telecommunications engineer of the coming decade will be called upon to design systems that serve not millions but **trillions of connected devices**, that operate not just in cities but in **every corner of the physical and digital world**, and that deliver not just voice but **immersive mixed reality, autonomous machine control, and real-time environmental intelligence**. The analytical frameworks, mathematical tools, architectural thinking, and measurement discipline developed in this Telephony module are your starting point for that career. They will serve you well.

A Final Word to the Student

Every time you make a phone call — whether from a landline, a smartphone, or a satellite handset — you are a participant in a global engineering system of staggering complexity and beauty. The signal carrying your voice has been sampled, encoded, multiplexed, transmitted over copper or fiber or radio waves, switched through digital exchanges and packet routers, decoded, and reproduced at the destination — all in less than 150 milliseconds. Behind every millisecond of that journey lies a century of engineering innovation, a library of international standards, and the professional expertise of hundreds of thousands of telecommunications engineers around the world. You are now one of them. Use this knowledge responsibly, creatively, and with the awareness that the systems you design and operate carry not just data, but the voices of human beings.

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