EEEN 464 – DIGITAL COMMUNICATION

STUDY GUIDE & REVISION NOTES ON

INTRODUCTION TO TELEPHONY

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1. BASIC CONCEPTS

- **Telephony**: Telephony is the technology associated with the electronic transmission of voice, fax, or other information between distant parties using systems historically associated with the telephone.

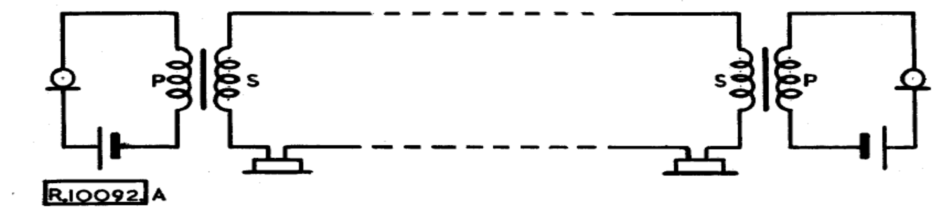

Figure 1. The Original concept of a telephone based on Alexander Bell Graham's invention in 1875.

- **Public Switched Telephone Network (PSTN):** Originally a global circuit-switched network for voice transmission, PSTN has evolved to a packet switched network capable of transmitting Data, images and videos.
- **Key Components of a Telephone Network**
    - **Subscriber Loop/Access Network**: Local connection between user and a telephone exchange traditionally referred to as local office. The term subscriber loop has been changing over time as telephony became digital and wireless where it is now referred to as the radio access network.
    - **Telephone exchange**: Switching centre connecting local subscribers or subscribers on other exchanges. Exchanges that switch traffic between exchanges are referred to as **transit exchanges**. Transit exchanges can be regional, national or international.
    - **Trunks**: High-capacity lines connecting telephone exchanges. The physical media for trunks can be copper wires, fibre-optic cables, terrestrial radio systems (microwave radio links) or satellite communication systems.

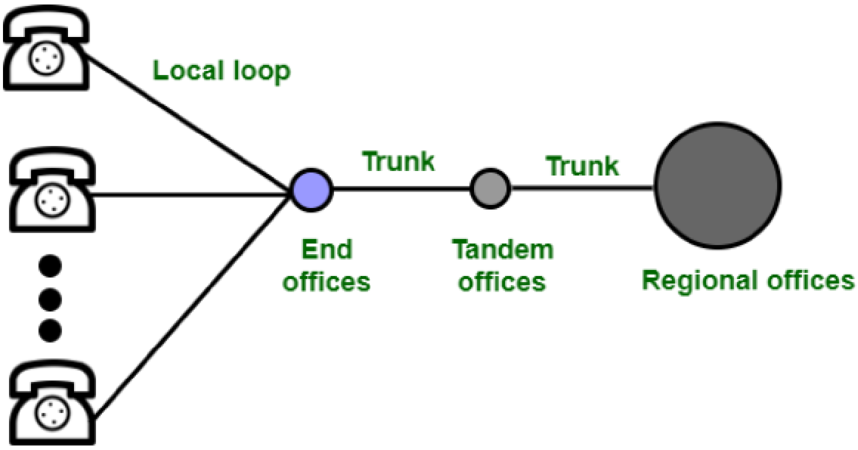

Figure 2. Components of an analogue telephone network.

2. ANALOG TELEPHONY

Analog telephony refers to the traditional method of voice communication using a fixed telephone line, where sound waves are converted into electrical signals that travel through copper wires or analogue radio systems to another phone. It's often referred to as Plain Old Telephone Service (POTS).

- **Voice Signal Characteristics**:
  - **Frequency range: 300 Hz – 3.4 kHz (bandlimited for efficiency).** This the international standard for analogue telephone voice. The frequency range was chosen because it carries most of the energy in a typical human voice, and it carries most of the voice interlegibility, i.e most human can interpret voice more clearly when voice is limited to this range. Further, it was necessary to limit range of voice in order to for modulators to function well.
  - **Transmitted as varying voltage on copper lines or modulating a carrier when telephone signals** are carried over terrestrial radio systems or satellite communication systems.
- **Subscriber Loop**
  - **2-wire circuit (Tx/Rx combined):** Traditional telephony used only two wires. Four wire communication (separate TX and RX) became necessary when radio communication and PCM networks came into existence.
  - **DC voltage (48V) powers the phone; ringing voltage (~90V AC).** Early telephone exchanges used -48V. This tradition has been maintained in many modern communication systems.
- **Call Setup**:
  - **Off-hook**: Loop current flow → dial tone.
  - **Pulse/Tone Dialling**: DTMF (Dual-Tone Multi-Frequency) for digit entry.

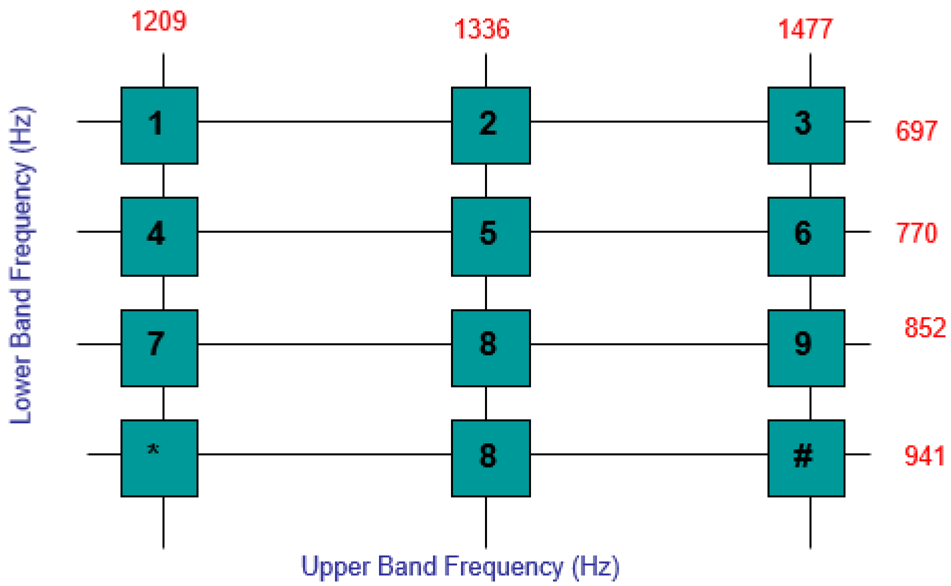

Figure 3. How a DTMF phone combines two frequencies to send digits over a telephone line.

3. DIGITAL TELEPHONY

Digital telephony uses of digital techniques to transmit voice and other communication signals over digital networks, like the internet. The key techniques used in digital telephony are:

a) **Analog-to-Digital Conversion**:

- **Sampling**: Nyquist rate (8 kHz for voice).
- **Quantization**: 8-bit samples → 64 kbps/channel (G.711 PCM).

b) **TDM (Time-Division Multiplexing)**:

- **DS0**: 64 kbps channel (1 voice call).
- **T1 (DS1)**: 24 DS0 channels + framing (1.544 Mbps) used in USA and Canada.
- **E1**: 30 DS0 channels (2.048 Mbps) – used in the rest of the world.

4. SIGNALLING SYSTEMS

a) **In-Band signalling**: Signalling information such as dialled telephone number are transmitted in the same channel or frequency-band as that used by the data/voice it is setting up/controlling.

- Example of in-band signalling is the **Dial-Tone Multifrequency (DTMF)** telephone which sends two-tones representing a dialled number within voice band.

b) **Out-of-Band signalling**:

- **Signalling System 7 (SS7)** is a set of telecommunication protocols that handle call setup, routing, and other control functions in the public switched telephone network (PSTN). It operates as an out-of-band signalling system, meaning signalling information is transmitted on separate data channels rather than within the voice path itself.

- Packet-based network for call control (setup/teardown).
- Key components: **SCP** (Service Control Point), **STP** (Signal Transfer Point).

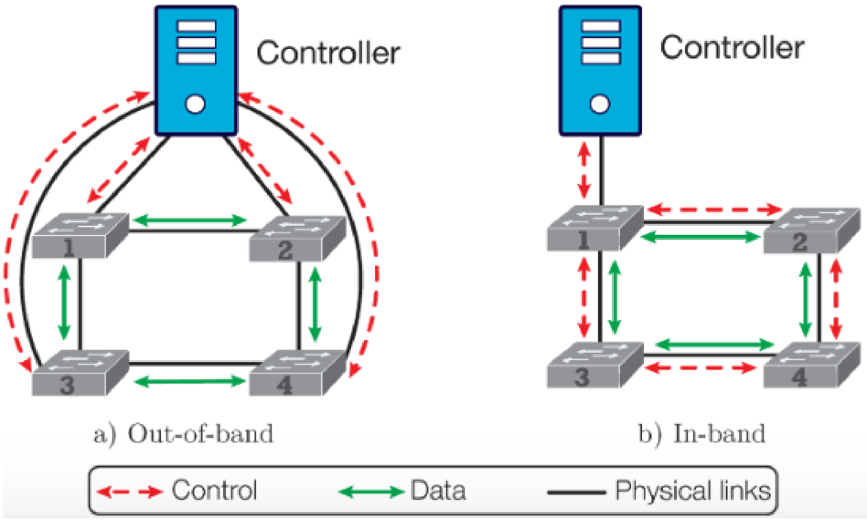

Figure 3. In-band and out-band signalling

5. SWITCHING TECHNIQUES

- **Circuit Switching**:
  - Dedicated path for entire call (e.g., PSTN).
  - Low latency but inefficient for bursty data.
- **Packet Switching**:
  - Voice broken into packets (used in VoIP).
  - Efficient but requires QoS management.

6. MODERN TELEPHONY

- **VoIP (Voice over IP)**:
  - Protocols: **SIP** (Session Initiation Protocol), **RTP** (Real-time Transport Protocol).
  - **Codecs**: G.711 (64 kbps), G.729 (8 kbps), Opus (adaptive).
- **Mobile Telephony**:
  - Generations: 2G (GSM), 3G (UMTS), 4G (LTE), 5G (VoNR).
  - **IMS (IP Multimedia Subsystem)**: SIP-based core for VoIP in 4G/5G.

7. KEY METRICS

- **QoS (Quality of Service)**:
  - **Latency**: <150 ms acceptable for voice.
  - **Jitter**: Variation in packet arrival (buffered at receiver).
  - **Packet Loss**: <1% tolerable.
- **MOS (Mean Opinion Score)**:
  - Voice quality rating (1–5; 4.0+ = "toll quality").

8. CHALLENGES

- **Echo**: Caused by impedance mismatches (solved with echo cancellers).
- **Delay**: Propagation delay + packetization (QoS critical). Delay in communication systems is usually specified as latency and/or jitter.
- **Security**: Eavesdropping (mitigated by **SRTP** – Secure RTP).

9. EVOLUTION & TRENDS

- **PSTN → VoIP**: Migration to IP-based networks (cost/efficiency).
- **5G Impact**: Ultra-reliable low-latency communication (URLLC) for voice.
- **WebRTC**: Browser-based real-time communication (no plugins).

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10. KEY FORMULAS

1. **Nyquist Sampling Rate**: fs≥2fmax*fs*≥2*f*max (e.g., 8 kHz for voice).
2. **Voice Bitrate (PCM)**: 8 bits/sample×8000 samples/sec=64 kbps8 bits/sample×8000 samples/sec=64 kbps.
3. **T1/E1/SONET/SDH Frame Duration**: 125 μs125 *μ*s (1 sample/channel/frame).