RQF LEVEL 3

NITVI301

NETWORKING AND INTERNET TECHNOLOGIES

VOIP System Installation

TRAINEE'S MANUAL

October, 2024

VOIP SYSTEM INSTALLATION

KOICA Korea International Cooperation Agency

TQUM TVET Quality Management Project

2024

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COORDINATION TEAM

RWAMASIRABO Aimable

MARIA Bernadette M. Ramos

MUTIJIMA Asher Emmanuel

PRODUCTION TEAM

Authoring and Review

NKUNDIBIZA Viateur

MUSABYIMANA Solange

Validation

NDAYISHIMIYE Eric

NIYONZIMA Louis

Conception, Adaptation and Editorial works

HATEGEKIMANA Olivier

GANZA Jean Francois Regis

HARELIMANA Wilson

NZABIRINDA Aimable

DUKUZIMANA Therese

NIYONKURU Sylvestre

MANIRAKIZA Jean de Dieu

Formatting, Graphics, Illustrations, and infographics

YEONWOO Choe

SUA Lim

SAEM Lee

SOYEON Kim

WONYEONG Jeong

MANIRAKORA Alexis

Financial and Technical support

KOICA through TQUM Project

TABLE OF CONTENT

ACRONYMS

API: Application Programming Interface

ATA: Analog Telephone Adapter

CLI: Command Line Interface

DHCP: Dynamic Host Configuration Protocol

DHCP: Dynamic Host Configuration Protocol

DID: Direct Inward Dialling

DNS: Domain Name System

DPI: Deep Packet Inspection

DSL: Digital Subscriber Line

DSLAM: Digital Subscriber Line Access Multiplexer

G.711: A standard for audio codec

G.729: Another standard for audio codec

GUI: Graphical User Interface

H.323: A standard for multimedia communications over packet-based networks

IAX: Inter-Asterisk eXchange

IP PBX: Internet Protocol Private Branch Exchange

IP: Internet Protocol

ISP: Internet Service Provider

IVR: Interactive Voice Response

LAN: Local Area Network

MAC: Media Access Control

MGCP: Media Gateway Control Protocol

NAT: Network Address Translation

PBX: Private Branch Exchange

PoE: Power over Ethernet

PSTN: Public Switched Telephone Network

QoS: Quality of Service

RTB: Rwanda TVET Board

RTP: Real-time Transport Protocol

SBC: Session Border Controller

SIP: Session Initiation Protocol

SLA: Service Level Agreement

SRTP: Secure Real-time Transport Protocol

T3/E3 Carrier: High-bandwidth internet connection

TCP: Transmission Control Protocol

TQUM Project: TVET Quality Management Project

TSS: Technical Secondary School

TVET: Technical and Vocational Education and Training

UDP: User Datagram Protocol

VLAN: Virtual Local Area Network

VoIP: Voice over Internet Protocol

VPN: Virtual Private Network

WAN: Wide Area Network

INTRODUCTION

This trainee's manual includes all the knowledge and skills required in Networking and Internet Technologies specifically for the module of **"VoIP system Installation".** Trainees enrolled in this module will engage in practical activities designed to develop and enhance their competencies. The development of this training manual followed the Competency-Based Training and Assessment (CBT/A) approach, offering ample practical opportunities that mirror real-life situations.

The trainee's manual is organized into Learning Outcomes, which is broken down into indicative content that includes both theoretical and practical activities. It provides detailed information on the key competencies required for each learning outcome, along with the objectives to be achieved.

As a trainee, you will start by addressing questions related to the activities, which are designed to foster critical thinking and guide you towards practical applications in the labor market. The manual also provides essential information, including learning hours, required materials, and key tasks to complete throughout the learning process.

All activities included in this training manual are designed to facilitate both individual and group work. After completing the activities, you will conduct a formative assessment, referred to as the end learning outcome assessment. Ensure that you thoroughly review the key readings and the 'Points to Remember' section.

MODULE CODE AND TITLE: NITVI301 VOIP SYSTEM INSTALLATION

Learning Outcome 1: Plan VOIP system installation
Learning Outcome 2: Deploy VOIP System equipment.
Learning Outcome 3: Configure VOIP system.
Learning Outcome 4: Operate VOIP system.
Learning Outcome 5: Test VOIP system
Learning Outcome 6: Maintain VOIP system.

Learning Outcome 1: Plan VOIP system installation

Indicative contents
1.1 Introduction to VoIP system **1.2 Identification of VoIP system installation requirements** **1.3 Design of VoIP system diagram** **1.4 Estimation of VoIP system cost**

Key Competencies for Learning Outcome 1: Plan VOIP system installation

Knowledge	Skills	Attitudes
● Description of VoIP system ● Identification of VoIP system diagram ● Description of site survey ● Identification of tools, materials and equipment ● Description of VoIP system services and network ● Identification of VoIP system cost	● Conducting site survey of VoIP system installation ● Selecting tool, materials and equipment in VoIP system installation ● Designing of VoIP system diagram ● Making the Cost estimation of VoIP System.	● Having teamwork ● Being Attentive ● Being Accurate ● Being confident. ● Being a critical thinker. ● Being analytical

Duration: 10 hrs

Learning outcome 1 objectives:

By the end of the learning outcome, the trainees will be able to:

1. Describe correctly VoIP system based on VoIP system requirements.
2. Describe properly VOIP System installation requirement based on Site survey findings.
3. Identify properly tools, Materials and equipment based on VoIP system requirements.
4. Describe correctly VoIP system services and network based on system requirements.
5. Design correctly VOIP diagram based on VoIP system requirements.
6. Estimate properly cost as per VoIP design.

Resources

Equipment	Tools	Materials
• Router • PBX • APBX • Phone • Switch • Computer	• Designing tools (Packet Tracer, eDraw max, GNS3) • Networking tool kit • Calculator • Tapes measures	• Internet bundles • Network cable • Cleaning solution • Cable connector

Indicative content 1.1: Introduction to VoIP System

Duration: 2 hrs

Theoretical Activity 1.1.1: Description of VOIP System

Tasks:

Task 1: Answer the following questions related to the description of VOIP System

i. What is VOIP System?

ii. Differentiate the types of VOIP System

iii. What is the applications area/use of VOIP system in real life?

iv. Explain the benefit of VOIP System

v. Outline the element of VOIP System

vi. Explain the working principle of VoIP system?

vii. Outline the VoIP system diagrams

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to the whole class

Task 4: Ask for clarification if any according to the expert presentation from trainer

Task 5: For more clarification, read the **key readings 1.1.1**

Key readings 1.1.1: Description of VOIP System

- **Definition**

VoIP stands for **"Voice over Internet Protocol**," and it refers to a technology that enables voice communication and multimedia sessions over the Internet or other IP-based networks. In simpler terms, VoIP allows you to make phone calls using the internet instead of traditional telephone lines. This technology converts your voice into digital data packets, which are then transmitted over the internet and reconverted into audio on the receiving end.

- **VOIP system Types and examples**

VoIP systems come in various types, each catering to specific needs and use cases.
Here are some common types of VoIP systems along with examples:

✓ **Residential VoIP Services**: Examples: Skype, Google Voice, Vonage, MagicJack

✓ **Business VoIP Services**: Examples: RingCentral, 8x8, Nextiva, Jive, Cisco Webex Calling

- ✓ **On-Premises VoIP Systems**: These are VoIP systems that are hosted and managed on-site by an organization. Examples: Asterisk, FreePBX, Cisco Unified Communications Manager
- ✓ **Hosted VoIP Systems**: Also known as cloud-based VoIP, these systems are hosted and maintained by a third-party provider. Examples: Grasshopper, Dialpad, Ooma Office, VirtualPBX
- ✓ **Mobile VoIP Applications**: These are apps that enable VoIP calling on mobile devices. Examples: WhatsApp, Viber, FaceTime, Facebook Messenger
- ✓ **Softphones**: Softphones are software applications that enable VoIP calls on computers or mobile devices. Examples: Zoiper, X-Lite, Bria, Linphone
- ✓ **IP-PBX Systems**: IP-PBX (Internet Protocol Private Branch Exchange) systems are used by businesses to manage internal and external calls.
  Examples: 3CX, Grandstream UCM Series, Yeastar S-Series
- ✓ **VoIP Gateways**: VoIP gateways connect traditional telephone networks (PSTN) to VoIP networks. Examples: Cisco VG Series Gateways, Patton SmartNode Gateways
- ✓ **Enterprise VoIP Solutions**: These are comprehensive VoIP solutions tailored for large organizations. Examples: Avaya Aura, Microsoft Teams Voice, Mitel MiVoice
- ✓ **Wholesale VoIP Services**: Wholesale VoIP services cater to businesses that resell VoIP services to end users. Examples: VoIP Innovations, Twilio, Nexmo (now part of Vonage)
- ✓ **SIP Trunking Services**: SIP (Session Initiation Protocol) trunking services enable businesses to make external calls using their IP-PBX systems. Examples: Twilio Elastic SIP Trunking, Flowroute, Bandwidth.

- ✓ **Open-Source VoIP Systems**: These are VoIP systems with open-source code that can be customized and extended. Examples: Asterisk, FreeSWITCH, Kamailio

- **Application area/use**
- ✓ **Personal Use**

  Online Communication: Popular apps like WhatsApp, Skype, and FaceTime use VoIP for voice calls, video calls, and messaging.

  Long-Distance Calls: VoIP offers significantly cheaper rates for international and long-distance calls compared to traditional phone services.

  Mobile Apps: Many smartphone apps utilize VoIP for voice and video calls, allowing users to communicate without incurring traditional phone charges.

- **Business Use**

- **Unified Communications:** VoIP integrates voice calls, video conferencing, instant messaging, and other communication tools into a single platform, enhancing collaboration and productivity.

- **Cost Reduction:** By eliminating the need for traditional phone lines, businesses can significantly reduce communication costs.
- **Remote Work:** VoIP enables employees to work from anywhere with an internet connection, facilitating remote work and business continuity.
- **Call Center Operations:** VoIP-based call centers offer features like call routing, automatic call distribution, and call recording, improving customer service efficiency.
- **Virtual Offices:** VoIP allows businesses to maintain a virtual presence with a professional phone system without the need for a physical office.

✓ **Other Applications**

- **IP PBX Systems:** VoIP technology powers modern IP PBX systems, offering advanced features and flexibility for businesses of all sizes.
- **Video Conferencing:** Platforms like Zoom and Google Meet use VoIP for audio transmission, enabling high-quality video conferencing experiences.
- **IoT Applications:** VoIP can be integrated into IoT devices for voice-controlled systems and remote monitoring.
- **Emergency Services:** Some regions are exploring the use of VoIP for emergency calls, providing alternative communication options in case of traditional network failures.

- **VoIP system elements.**

  A VoIP (Voice over Internet Protocol) system is composed of several key elements that work together to enable voice communication over the internet or IP-based networks. Each element plays a specific role in the process of transmitting and receiving voice data.

✓ **Endpoints**: Endpoints are the devices used to initiate and receive VoIP calls. These can include computers, smartphones, VoIP phones, tablets, and even traditional phones with VoIP adapters. Endpoints encode analogy audio signals into digital data packets for transmission and decode received packets back into audio.

✓ **VoIP Software**: VoIP software is responsible for managing the encoding, transmission, and decoding of voice data. It may be embedded in communication applications, operating systems, or dedicated VoIP hardware. This software handles tasks like packetization, error correction, and jitter buffering to ensure smooth communication.

✓ **Codecs**: Codecs (coder-decoder) are algorithms used to compress and decompress audio data. They determine how audio is transformed into digital data and back. Codecs affect call quality, bandwidth usage, and latency. Common codecs include G.711, G.729, and Opus.

- ✓ **VoIP Server**: The VoIP server is a central component that manages various functions within the VoIP system. It handles call routing, signalling, user authentication, and more. Session Initiation Protocol (SIP) servers are common in VoIP systems to establish and control calls.
- ✓ **Gateways**: VoIP gateways act as bridges between VoIP networks and traditional PSTN (Public Switched Telephone Network) systems. They convert VoIP data into formats suitable for traditional phone networks and vice versa.
- ✓ **Network Infrastructure**: A reliable and stable internet connection is essential for VoIP communication. Quality of Service (QoS) mechanisms prioritize VoIP traffic to ensure consistent call quality. Low latency, minimal jitter, and sufficient bandwidth are critical to maintaining clear conversations.
- ✓ **Protocols**: VoIP systems rely on protocols for communication and signalling between devices and servers. SIP (Session Initiation Protocol) is a common protocol for initiating, modifying, and terminating VoIP sessions. RTP (Real-time Transport Protocol) is used for transmitting audio and video data.
- ✓ **Firewalls and NAT Traversal**: Firewalls and Network Address Translation (NAT) can hinder VoIP traffic due to their security mechanisms. VoIP systems often use techniques like STUN, TURN, and ICE to overcome these obstacles.

- ✓ **Security Measures**: VoIP systems need security mechanisms to protect against eavesdropping, unauthorized access, and other threats. Encryption, secure authentication, and regular updates are crucial to maintaining a secure VoIP environment.
- ✓ **VoIP Providers**: VoIP providers offer services that enable individuals and businesses to use VoIP technology for voice communication. They provide the necessary infrastructure, connectivity, and often additional features like call recording and conferencing.

- **Benefits VoIP System**

VoIP (Voice over Internet Protocol) systems offer a wide range of benefits compared to traditional telephone systems.

- ✓ **Cost Savings**: VoIP calls are typically cheaper than traditional landline calls, especially for long-distance and international calls. This is particularly beneficial for businesses that need to communicate across different locations.
- ✓ **Lower Infrastructure Costs**: VoIP systems often require minimal physical infrastructure. They can be implemented using existing internet connections and devices, reducing the need for extensive cabling and dedicated phone lines.

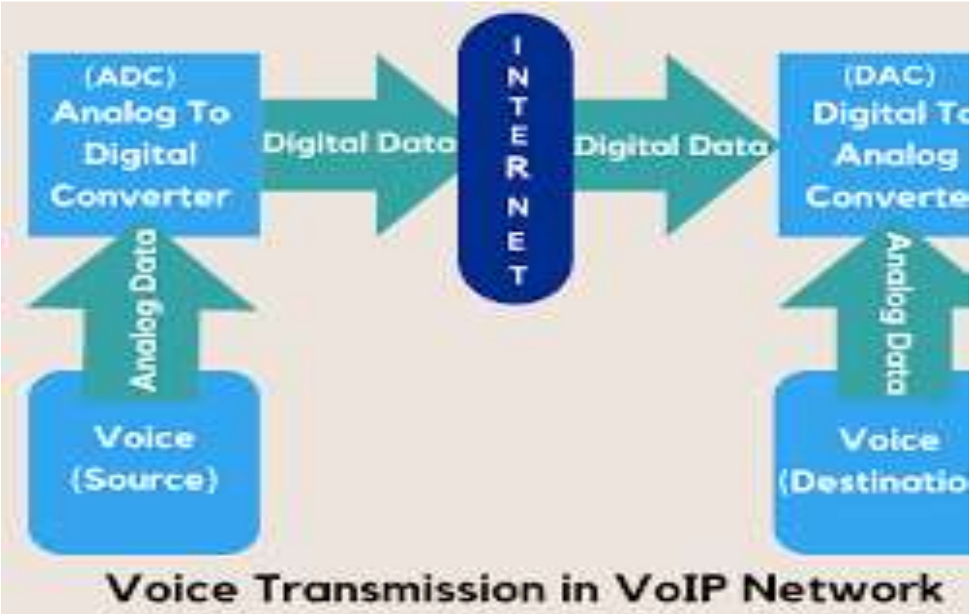
- ✓ **Flexibility and Mobility**: VoIP allows users to make and receive calls from anywhere with an internet connection. This is especially useful for remote workers, travellers, and businesses with distributed teams.
- ✓ **Scalability**: VoIP systems are easily scalable. Adding new users or lines typically involves minimal configuration and doesn't require physical line installations.
- ✓ **Rich Communication Features**: VoIP systems often come with advanced communication features such as call forwarding, voicemail-to-email, video conferencing, call recording, and virtual attendants.

- ✓ **Integration with Other Systems**: VoIP can be integrated with other communication and collaboration tools, such as email clients, customer relationship management (CRM) software, and unified communication platforms.
- ✓ **Global Reach**: VoIP eliminates geographical restrictions, making it possible to have local phone numbers in different countries without physical presence there.
- ✓ **High-Quality Audio**: VoIP technology has improved over the years, leading to better call quality. Many modern codecs and protocols ensure that voice data is transmitted with minimal delay and high fidelity.

- ✓ **Easy Management**: VoIP systems can be managed through web-based interfaces, allowing administrators to configure settings, add users, and monitor call activity from a central location.
- ✓ **Business Continuity**: In the event of natural disasters or other disruptions, VoIP systems can provide more reliable communication options, as long as there is internet connectivity.
- ✓ **Environmental Benefits**: VoIP reduces the need for extensive hardware and cabling, which can contribute to a decrease in electronic waste. Additionally, the energy consumption associated with VoIP is generally lower compared to traditional phone systems.

- ✓ **Unified Communication**: VoIP systems can integrate voice, video, and messaging into a single platform, creating a unified communication experience that enhances collaboration.
- ✓ **Number Portability**: VoIP systems often allow users to keep their existing phone numbers when switching from traditional phone services, making the transition smoother.
- ✓ **Real-time Analytics**: Many VoIP systems offer analytics and reporting features that provide insights into call patterns, call durations, and other useful metrics.

- **Working principle**



✓ **Analog-to-Digital Conversion**

When a user speaks into a microphone or handset, their voice is initially in analogy form. The VoIP system's endpoint (device) first converts this analogy voice signal into digital data using an analogy-to-digital converter (ADC). This digital representation breaks the audio into discrete samples.

✓ **Voice Compression**

The digital audio data is often compressed to reduce the amount of data that needs to be transmitted. Codecs (coder-decoders) are used for this purpose. Codecs encode the audio data, removing redundant or less perceptible information while maintaining acceptable audio quality.

✓ **Packetization**

The compressed audio data is divided into small packets. Each packet contains a portion of the audio data along with additional information, such as source and destination addresses, sequence numbers, and timestamps. The size and number of packets depend on factors like codec used and network conditions.

✓ **Signalling and Session Setup**

Before actual audio transmission can occur, a signalling protocol is used to establish a connection between the caller and the recipient. One of the most common signalling protocols is Session Initiation Protocol (SIP). During this phase, the calling party sends a SIP INVITE to the recipient's device, which indicates a desire to establish a communication session.

✓ **Packet Transmission**

The packets containing the digitized and compressed audio are sent over the IP-based network. These packets may traverse multiple routers and switches, potentially taking different routes to reach the destination.

- ✓ **Quality of Service (QoS)**

To ensure clear and reliable voice communication, VoIP systems often employ Quality of Service mechanisms. These prioritize VoIP traffic over other types of data to minimize latency, jitter, and packet loss, which can negatively impact call quality.

- ✓ **Packet Reception and Reordering**

At the receiving end, the packets are received, and their order is rearranged based on sequence numbers and timestamps. This process helps ensure that the audio is reconstructed in the correct order.

- ✓ **Decompression and Digital-to-Analog Conversion**

The received packets are then passed through a codec on the recipient's end, which decodes the compressed audio data. The result is a digital audio stream that's passed through a digital-to-analogy converter (DAC), which transforms the digital audio data back into analogy signals.

- ✓ **Audio Playback**

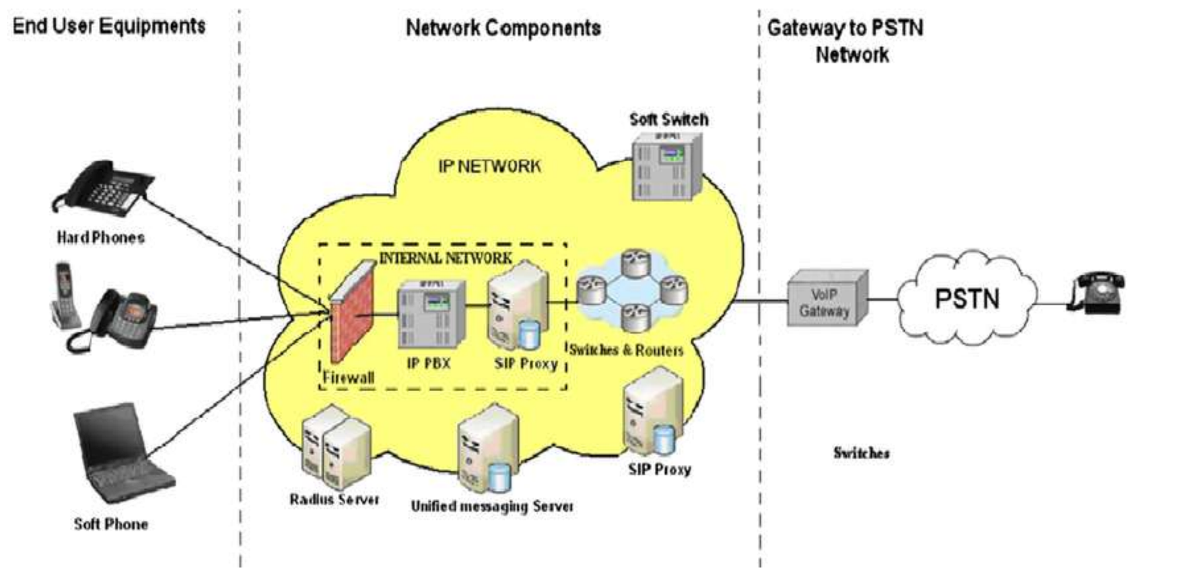
The analogy audio signals are sent to the user's speaker or headset for playback. This completes the process, allowing the recipient to hear the caller's voice.

- ✓ **Session Termination**

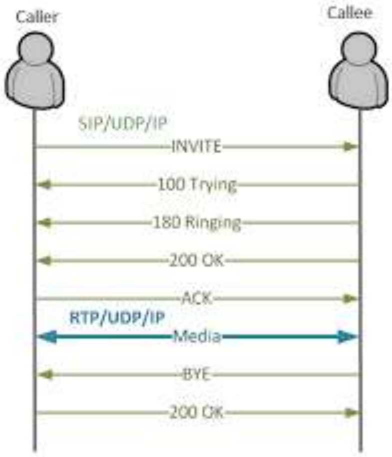
Once the communication is finished, signalling protocols are used to terminate the session. The devices exchange messages to indicate the end of the call, and any necessary resources are released.

- **VOIP system Diagram**
- ✓ **High-Level VoIP Architecture Diagram**

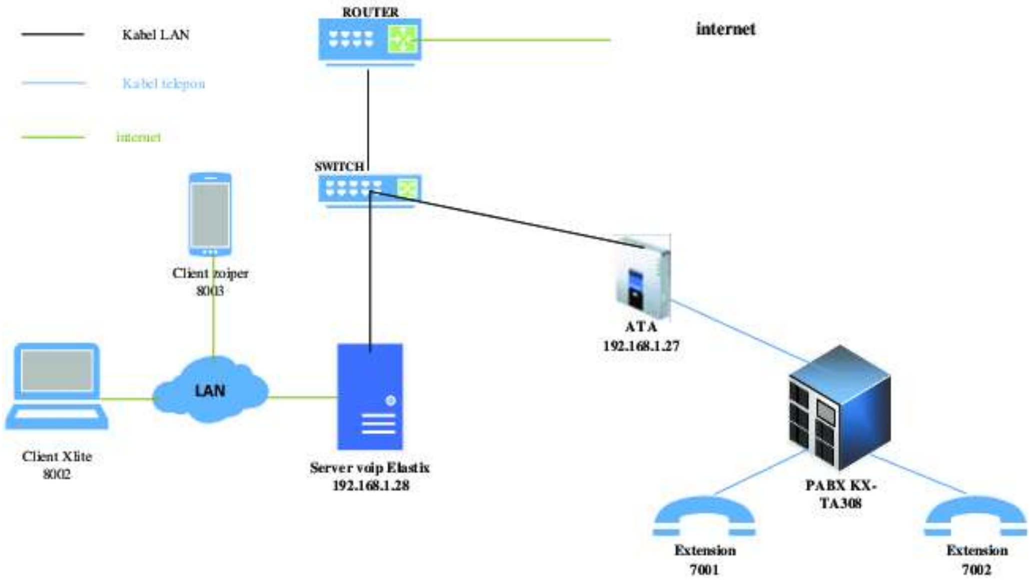
This diagram provides an overview of the major components in a VoIP system. It includes endpoints, VoIP servers, gateways, and the internet. Arrows indicate the flow of voice data packets and signalling messages between these components.



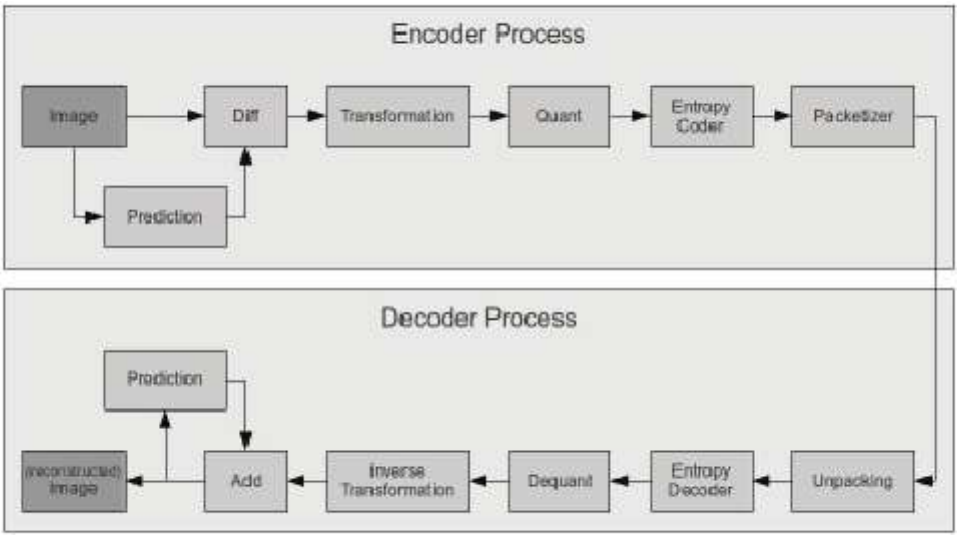
- ✓ **VoIP Call Flow Diagram**: A call flow diagram illustrates the sequence of steps and interactions that occur when establishing a VoIP call. It includes SIP signalling messages, RTP packet flow for voice data, and the involvement of different components such as SIP servers and user agents.



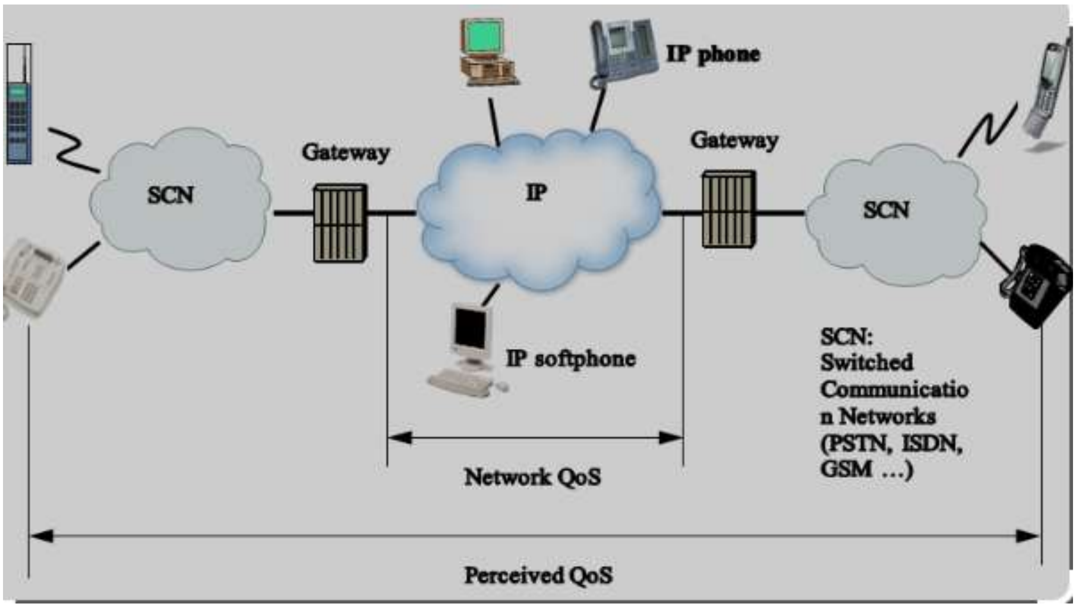
- ✓ **Network Topology Diagram**: This diagram illustrates how VoIP components are connected within a network. It showcases endpoints (computers, phones), VoIP servers (SIP servers, media servers), gateways, routers, and switches, as well as the internet connection.



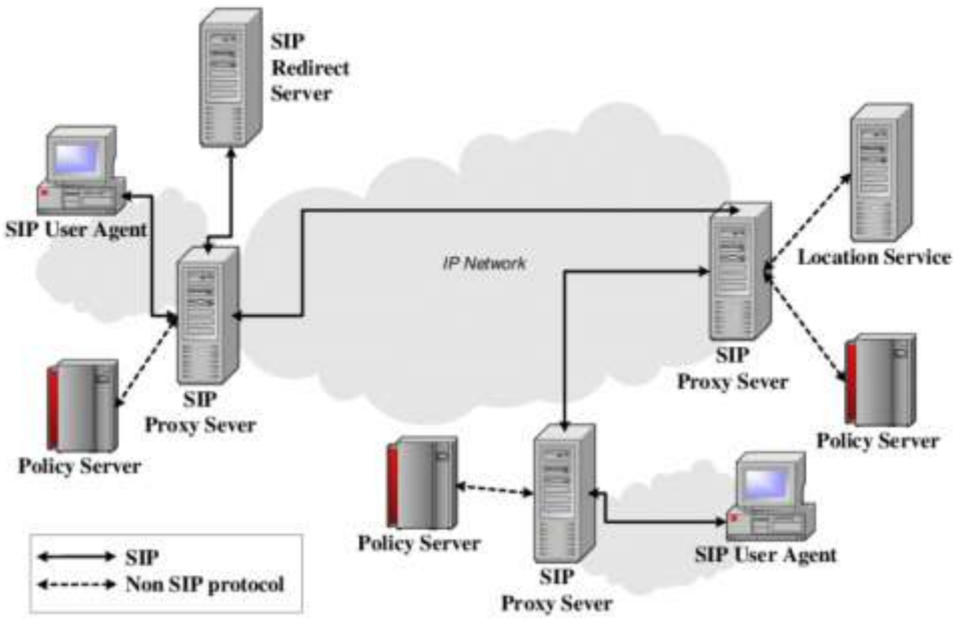
- ✓ **Codec Selection Diagram**: This diagram shows the process of choosing and applying a codec for compressing and decompressing audio data. It illustrates the trade-offs between bandwidth usage and call quality that different codecs offer.



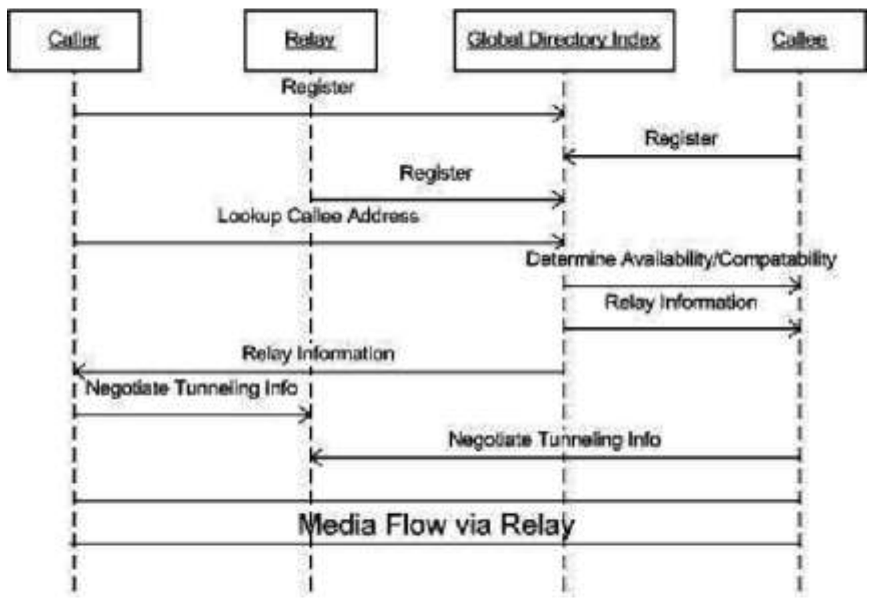
- ✓ **QoS Diagram**: Quality of Service (QoS) diagrams depict how VoIP traffic is prioritized within a network. They show how QoS mechanisms manage and ensure low latency, minimal jitter, and reduced packet loss for VoIP packets compared to other types of data.



- ✓ **VoIP Security Diagram**: This diagram focuses on the security aspects of a VoIP system. It illustrates encryption mechanisms, firewalls, intrusion detection systems, and the safeguarding of signalling and media traffic from unauthorized access.
- ✓ **SIP Server Configuration Diagram**: This diagram details the setup of a SIP server, including user registration, call routing, and session establishment. It can highlight how SIP headers and messages are used for call initiation and control.



- ✓ **VoIP Mobile App Interaction Diagram**: For mobile VoIP applications, this diagram shows the interaction between the app, SIP server, and network. It covers registration, call initiation, media transmission, and call termination.



Points to Remember

- ✓ VOIP (Voice over Internet Protocol) enables voice communication and multimedia sessions over the internet or IP networks.
- ✓ Types of VoIP include Hosted VOIP and On-Premises VOIP
- ✓ Applications of VoIP include Business Communication, Telecommuting, and Consumer Services.
- ✓ Benefits of VoIP system include Cost Savings, Flexibility, Scalability, Integration, and Advanced Features.
- ✓ Elements of VoIP consist of IP Phones/Softphones, VOIP Gateway, Session Border Controller (SBC), and VOIP Server.
- ✓ VOIP works by digitizing voice signals into data packets and involves endpoints, servers, gateways, and the internet in its diagram.

Application of learning 1.1.

You are a technical support representative for a VoIP service provider. A new customer has contacted you to inquire about the details of their VoIP system. They are interested in understanding how VoIP works, the components involved, and the benefits it offers compared to traditional phone systems.

Indicative content 1.2: Identification of VoIP System Installation Requirements

Duration: 2 hrs

Theoretical Activity 1.2.1: Description of evaluation of environment

Tasks:

Task 1: In small groups, answer the following questions:

i. What is physical site survey?
ii. Why physical site survey needed while deploying VoIP system?
iii. Provide any VoIP system blueprint.
iv. Elaborate tools, materials and equipment used in VoIP system?
v. Differentiate the VoIP Service?
vi. What is VoIP Network?

Task 2: Provide the answer for the asked questions and write them on papers.
Task 3: Present the findings/answers to the whole class
Task 4: Ask for clarification if any according to the expert presentation from trainer
Task 5: For more clarification, read the key readings 1.2.1.

Key readings 1.2.1.:

- **Evaluation of environment.**

✓ **Define physical site survey.**

A physical site survey refers to the systematic assessment of a location or premises to evaluate its suitability for deploying VoIP communication infrastructure. This survey is a crucial step in ensuring the successful implementation of VoIP services. It involves examining the existing network infrastructure, assessing the quality of cabling, and identifying potential issues that may affect VoIP performance, such as network congestion, interference, or insufficient power sources. Additionally, physical site surveys in VoIP often focus on the deployment of Voice over IP phones and the positioning of network switches, routers, and other essential equipment to optimize call quality, minimize latency, and ensure the overall reliability of the VoIP system. This thorough evaluation helps organizations make informed decisions about the necessary upgrades or modifications required to support VoIP technology effectively and deliver high-quality voice communication over the IP network.

- **Why physical site needed in VoIP system?**

A physical site survey is a crucial component when planning for the deployment of various systems and technologies, including network infrastructure, security systems, and more. Conducting a physical site survey helps ensure that the design and implementation meet the specific needs and constraints of a physical location.

The main components of a physical site survey typically include:

- **Site Information and Documentation**: Gather existing documentation, such as architectural plans, blueprints, and building layouts, to understand the physical layout of the site.
- **Site Access and Security**: Identify access points, security measures, and any restrictions that might affect the deployment.
- **Environmental Considerations**: Assess environmental factors such as temperature, humidity, and potential sources of interference (e.g., electromagnetic interference) that can impact equipment performance.
- **Power Infrastructure**: Evaluate the availability and quality of power sources, including the location of electrical outlets and the capacity to support the equipment being deployed.
- **Space and Layout**: Measure available space and assess how equipment will be physically positioned. This includes rack space for servers and network equipment.
- **Cabling Infrastructure**: Inspect existing cabling infrastructure (e.g., Ethernet, fibre optic, power, and phone lines) to determine its suitability for the new technology.
- **Network Connectivity**: Examine network connectivity options and identify the locations of network switches, routers, and data jacks. Determine if additional networking infrastructure is needed.
- **Wireless Coverage**: Conduct a wireless site survey to assess signal strength and potential sources of interference for Wi-Fi and other wireless technologies.
- **Physical Security Assessment**: Evaluate physical security measures, such as doors, locks, surveillance cameras, and access control systems.
- **Safety and Compliance**: Ensure compliance with safety regulations and standards, including fire safety, emergency exits, and accessibility for people with disabilities.

- **Heating, Ventilation, and Air Conditioning (HVAC)**: Review HVAC systems to ensure that they can adequately control temperature and humidity for equipment rooms.
- **Site Layout and Traffic Flow**: Analyse the flow of people and equipment within the site to ensure efficient operation and safety.
- **Equipment Placement**: Plan the precise locations for equipment, including servers, network switches, security cameras, and other hardware.
- **Future Expansion**: Consider future needs and expansion options when designing the layout and infrastructure.
- **Documentation and Reporting**: Create a detailed report that summarizes the findings, recommendations, and requirements identified during the site survey. This document will be used as a reference during the implementation phase.
- **Budget and Resource Planning**: Based on the survey results, develop a budget and resource plan for the deployment, including equipment procurement, labor, and other resources required.
- **Stakeholder Coordination**: Communicate the survey findings and plans with relevant stakeholders, including site owners, management, and the project team.

✓ **Discuss system usage.**

System usage in a Voice over Internet Protocol (VoIP) system refers to the utilization and consumption of resources and components within the VoIP infrastructure to facilitate the transmission of voice and multimedia data over IP networks. This encompasses various aspects, including the allocation of bandwidth, server capacity, and the overall efficiency of the VoIP system. System usage is essential for maintaining call quality, reliability, and optimal performance. Network administrators and VoIP engineers closely monitor system usage to ensure that the system operates within specified parameters.

System usage metrics in VoIP can include bandwidth consumption, call volume, concurrent call capacity, server CPU and memory utilization, as well as network latency and jitter. Effective management of system usage helps in the proactive identification and resolution of issues such as network congestion, call drops, or poor call quality. By monitoring and analysing these metrics, VoIP administrators can make informed decisions about network upgrades, resource allocation, and Quality of Service (QoS) settings to ensure a smooth and high-quality communication experience for users. Additionally, system usage data is valuable for capacity planning, allowing organizations to scale their VoIP infrastructure as their communication needs grow, and ensuring that the system continues to meet performance expectations.

- **Provide blueprint design.**

Blueprint design is a detailed plan or drawing that outlines the layout, dimensions, and specifications of a project. It's a crucial tool for architects, engineers, and builders, as it provides a clear and concise visual representation of the final product.

Key Components of a Blueprint Design

- **Floor Plans:** Detailed drawings of each floor, showing the layout of rooms, walls, doors, and windows.
- **Elevations:** Views of the building's exterior from different angles, showing the height, shape, and materials.
- **Sections:** Cross-sectional views of the building, revealing the internal structure and materials.
- **Details:** Close-up drawings of specific components, such as doors, windows, and fixtures.
- **Specifications:** Written descriptions of materials, finishes, and construction techniques.

Blueprint Design Process

- **Concept Development:** Create initial sketches and ideas for the project.
- **Site Analysis:** Assess the site's conditions, including topography, zoning regulations, and utility access.
- **Design Development:** Refine the design based on the concept and site analysis.
- **Blueprint Creation:** Develop detailed drawings and specifications.
- **Review and Approval:** Obtain necessary approvals from relevant authorities.
- **Construction Documentation:** Provide additional drawings and specifications for construction purposes.

Blueprint Design Software

- **AutoCAD:** A popular industry-standard software for creating 2D and 3D drawings.
- **Revit:** A Building Information Modelling (BIM) software that allows for integrated design and construction management.
- **SketchUp:** A user-friendly 3D modelling software for creating architectural designs and visualizations.
- **Vectorworks:** A versatile design software for architects, engineers, and landscape designers.

Importance of Blueprint Design

- **Clear Communication:** Blueprints provide a clear and concise visual representation of the project, ensuring everyone involved understands the design intent.
- **Accuracy and Precision:** Blueprints help to avoid errors and misunderstandings during construction.
- **Efficiency:** Blueprints can help streamline the construction process by providing a detailed plan to follow.

- **Cost Control:** Accurate blueprints can help to control costs by minimizing waste and rework.

- **VoIP services**

key factors to consider when identifying your VoIP system installation requirements:

- ✓ **Business Needs Assessment**

- **Number of users:** How many employees will use the VoIP system?
- **Call volume:** What is the average number of calls handled daily?
- **Call features:** What features are essential (e.g., call forwarding, voicemail, call conferencing, auto-attendant)?
- **Integration:** Do you need integration with other business systems (CRM, email, etc.)?
- **Scalability:** Will your business grow? Consider a system that can accommodate future expansion.
- **Reliability:** What is your tolerance for service interruptions?
- **Security:** What level of security is required for your data and communications?

- ✓ **Network Assessment**

- **Internet speed:** Ensure sufficient bandwidth for voice traffic.
- **Network infrastructure:** Evaluate your existing network's capacity to handle VoIP.
- **Wi-Fi coverage:** If using wireless phones, assess Wi-Fi strength and coverage.
- **Network security:** Implement firewalls and other security measures to protect your system.

- ✓ **Hardware and Software Requirements**

- **VoIP phones:** Determine if you need desk phones, softphones, or a combination.
- **PBX system:** Decide between a cloud-based or on-premises PBX.
- **Additional equipment:** Consider headsets, adapters, and other necessary hardware.
- **Software applications:** Evaluate integration needs with CRM, email, and other software.

- ✓ **Budget and ROI**

- **Cost analysis:** Compare the total cost of ownership for VoIP versus traditional phone systems.
- **Return on investment:** Calculate potential savings on long-distance calls, equipment, and maintenance.

- ✓ **Service Provider Selection**

- **Features and pricing:** Compare different VoIP service providers based on your needs.
- **Reliability and customer support:** Consider the provider's reputation and support options.
- **Scalability:** Ensure the provider can accommodate your business growth.

- ✓ **Implementation and Training**

- **Planning and deployment:** Develop a detailed implementation plan.
- **User training:** Provide comprehensive training to employees on using the new system.

- ✓ **Additional Considerations**

- **Emergency services:** Ensure compatibility with your local emergency services.
- **Regulatory compliance:** Adhere to industry-specific regulations (e.g., HIPAA for healthcare).
- **Disaster recovery:** Develop a plan for system backup and recovery.

- **VoIP network**

VoIP network refers to the infrastructure that enables voice communication over the internet. It's essentially a digital pathway for voice data to travel between users. This contrasts with traditional phone systems that rely on physical copper wires.

- ✓ **How Does it Work?**

1. **Analog to Digital Conversion:** Your voice is converted into digital data.
2. **Packet Creation:** The digital voice data is divided into small packets.
3. **Transmission:** These packets are sent over the internet to the recipient.
4. **Reassembly:** At the receiving end, the packets are reassembled and converted back into analog sound.

- ✓ **Components of a VoIP Network**

- **Endpoints:** These can be VoIP phones, softphones (software on computers or smartphones), or analog phones connected to an Analog Telephone Adapter (ATA).
- **PBX (Private Branch Exchange):** This is the central control unit that manages calls within a network. It can be on-premises or cloud-based.
- **Media Gateway:** Converts analog signals to digital and vice versa for interoperability with the traditional phone network.

- **Internet Connection:** The backbone for transmitting voice data.

✓ **Types of VoIP Networks**

- **Public VoIP:** Uses the public internet for call transmission.
- **Private VoIP:** Uses a private network within an organization.

✓ **Advantages of VoIP Networks**

- **Cost-effective:** Lower call charges, especially for long-distance and international calls.
- **Flexibility:** Can be used on various devices (computers, smartphones, tablets).
- **Scalability:** Easy to add or remove users.
- **Additional Features:** Offers features like call forwarding, voicemail, conference calling, and more.

✓ **Challenges of VoIP Networks**

- **Quality of Service (QoS):** Requires sufficient bandwidth and network stability for clear calls.
- **Security:** Vulnerable to hacking and eavesdropping if not properly secured.
- **Emergency Calls:** May have limitations in some areas.

- **VoIP system tools**

✓ Designing tools (Packet Tracer, eDraw max, GNS3)

1. **Packet Tracer**

Packet Tracer is a network simulation tool developed by Cisco Systems. It's primarily used for teaching and learning networking concepts and practicing network configurations.

- **Key Features**:
  - Simulate network topologies and Cisco devices (routers, switches, etc.).
  - Allows you to create, configure, and test network setups without physical hardware.
  - Ideal for educational purposes and preparing for Cisco certifications.
  - Includes a variety of networking components and protocols.

2. **eDraw Max**

eDraw Max is a diagramming software used for creating diagrams, flowcharts, mind maps, and various other visual representations of ideas and processes.

- **Key Features**:
  - Offers a wide range of templates and symbols for diagram creation.
  - Supports a variety of diagram types, including flowcharts, organizational charts, floor plans, and more.
  - Provides a user-friendly drag-and-drop interface.
  - Ideal for professionals and teams needing to create visual content for documentation, presentations, and planning.

3. **GNS3 (Graphical Network Simulator-3)**

GNS3 is a network emulator that is used for network design and simulation. It's particularly popular among network professionals for testing and verifying complex network configurations.

- **Key Features**:
  - Emulates a variety of networking devices, including Cisco, Juniper, and more.
  - Allows you to create complex network topologies.
  - Supports integration with real hardware and virtual machines for realistic testing.
  - Useful for network engineers and administrators to design and test network solutions.

Each of these tools has its unique purpose and strengths. Choose the one that aligns with your specific needs:

- If you're learning or teaching networking Packet Tracer is an excellent choice.
- For creating diagrams and visual representations of information, eDraw Max is a versatile option.
- Network professionals and administrators may prefer GNS3 for network design and testing.

✓ **Networking tool kit**

A networking toolkit, often referred to as a network toolkit or network administrator's toolkit, is a collection of software and hardware tools used by network professionals and administrators to manage, maintain, troubleshoot, and optimize computer networks.

These toolkits typically include a variety of utilities and devices to assist in tasks related to network monitoring, configuration, security, and diagnostics. The specific tools and equipment in a network toolkit may vary depending on the network administrator's needs and the complexity of the network being managed.

A networking toolkit may include, but is not limited to, the following components:

1. **Software Tools**:
   - **Network Monitoring Tools**: Software for monitoring network performance, traffic, and device status.
   - **Network Configuration Tools**: Utilities for configuring and managing network devices, such as routers and switches.
   - **Security Tools**: Software to detect and respond to security threats, including firewalls, intrusion detection systems, and antivirus software.
   - **Diagnostics Tools**: Tools for troubleshooting network issues, including ping, traceroute, and network analyzers.
   - **Remote Administration Software**: Tools for remotely managing network devices and servers.
2. **Hardware Tools**:
   - **Network Cable Testers**: Devices for testing and diagnosing network cable connections and integrity.
   - **Crimping and Cable Termination Tools**: Equipment for terminating and crimping network cables.
   - **Network Analyzers**: Hardware tools for analyzing network traffic, identifying issues, and optimizing network performance.
   - **Ethernet Switches and Routers**: Hardware devices used in network configurations and testing.
   - **Patch Panels**: Used for organizing and connecting network cables in data centers.
3. **Diagnostic Equipment**:
   - **Tone and Probe Kit**: Used for tracing and locating network cables.
   - **Loopback Adapters**: Devices for testing network interfaces and connections.
   - **WiFi Analyzer**: Tools for analyzing and optimizing wireless network performance.

- **Network Cable Tester**: Devices for verifying cable connectivity and detecting cable faults.

4. **Tool Bags and Cases**: Carrying cases or bags designed to organize and transport network tools and equipment.
5. **Documentation and Labeling Tools**: Label makers and tools for documenting and labeling network components.

✓ **Calculator**

Calculators can play a role in VoIP (Voice over Internet Protocol) systems in various ways, particularly in the planning, provisioning, and troubleshooting of VoIP networks. There are some specific uses of calculators in VoIP systems:

1. **Bandwidth Calculator**: VoIP calls require a certain amount of bandwidth to transmit voice data over the internet. Network administrators and VoIP service providers use bandwidth calculators to estimate the amount of bandwidth required for a given number of concurrent calls. This helps in capacity planning and ensuring that the network can handle the expected call volume without degradation in call quality.
2. **Codec Selection Calculator**: Different audio codecs are used in VoIP systems, and the choice of codec can impact call quality and bandwidth requirements. Codec selection calculators assist in determining the most suitable codec based on factors like available bandwidth, desired call quality, and network conditions.
3. **Jitter and Latency Calculators**: VoIP calls are sensitive to jitter (variation in packet arrival times) and latency (packet travel time). Calculators can help assess and predict the impact of jitter and latency on call quality, allowing network administrators to make adjustments as needed to minimize disruptions during calls.
4. **Subnet and IP Address Calculations**: Proper IP addressing and subnetting are crucial in VoIP systems. Calculators assist in IP address management, subnet design, and allocation of addresses for VoIP devices, ensuring that IP conflicts are avoided and network routing is efficient.
5. **Quality of Service (QoS) Calculations**: QoS is vital for VoIP to prioritize voice traffic over other types of data. Calculators help determine the necessary QoS settings, such as DSCP (Differentiated Services Code Point) values and traffic shaping parameters, to ensure that voice packets are treated with the appropriate level of priority.
6. **Call Volume Estimation**: Calculators can help estimate call volume, call duration, and peak call times. This information is valuable for provisioning network resources and sizing the VoIP infrastructure accordingly.

7. **Cost Calculations**: For businesses and service providers, cost calculators can help in estimating the cost of operating a VoIP system. This includes factors like the cost of bandwidth, equipment, licenses, and maintenance.
8. **VoIP Codecs Bitrate Calculators**: These calculators determine the bit rate for different VoIP codecs, which can be useful for understanding bandwidth requirements and network capacity planning.
9. **Echo Delay Calculations**: Echo can be a common issue in VoIP systems. Calculators can help determine the maximum acceptable echo delay, which is essential for echo cancellation and echo suppression settings.
10. **Call Setup Time Calculations**: Calculators can estimate the time it takes to establish a VoIP call, considering factors like call signaling and session initiation. This is important for ensuring efficient call setup and a good user experience.

- ✓ **Tapes measures**

The use of a tape measure is more commonly associated with physical aspects of installation, such as mounting or positioning hardware, determining cable lengths, or planning the layout of equipment. Here are a few scenarios where a tape measure might be used during a VoIP system installation:

1. **Cable Length Calculation:** When installing Ethernet cables for connecting VoIP phones or other network devices, you might use a tape measure to determine the appropriate cable lengths to ensure a neat and organized installation.
2. **Mounting Hardware:** If you're mounting network equipment, phones, or other hardware on walls, racks, or other structures, you may use a tape measure to ensure proper spacing, alignment, and placement.
3. **Cable Management:** Using a tape measure to plan cable routing and cable management can help maintain a clean and efficient installation. This is important for avoiding cable clutter and ensuring easy access for maintenance.

4. **Spacing and Layout:** In situations where multiple network devices are being installed in a specific layout, a tape measure can help ensure that everything is positioned correctly and that there's enough space for proper ventilation and maintenance access.

- **VoIP System materials**

✓ **Internet bundles**

Internet bundles typically refer to packages or plans offered by internet service providers (ISPs) that combine multiple services or features into a single package for a specific price. These bundles often include various components such as:

1. **Internet Access**: The primary component is internet service, which can be offered in different speed tiers (e.g., 10 Mbps, 100 Mbps, 1 Gbps) depending on the plan.
2. **Cable or Satellite TV**: Some bundles include access to cable or satellite television services. This can include a variety of channels, including premium channels like HBO or sports packages.
3. **Home Phone**: Bundles may also include a home phone service with features such as unlimited calling or international calling options.
4. **Streaming Services**: In some cases, ISPs partner with streaming services like Netflix or Amazon Prime, and these services are included in the bundle.
5. **Home Security**: Some bundles include home security services, such as smart locks, cameras, and monitoring services.
6. **Wireless Services**: Some ISPs offer mobile or wireless services as part of their bundles, including data plans and cell phone service.

✓ **Network cable**

A network cable is a type of cable used to connect devices in a computer network. It serves as a physical medium through which data is transmitted between devices, such as computers, routers, switches, and other networking equipment. Network cables are essential for establishing wired connections in both local area networks (LANs) and wide area networks (WANs). There are several types of network cables, each with its own characteristics and use cases. Some common types of network cables include:

1. **Ethernet Cable (Cat5e, Cat6, Cat6a, Cat7, Cat8)**: Ethernet cables are the most widely used network cables for wired network connections. They come in various categories, such as Cat5e, Cat6, Cat6a, Cat7, and Cat8, with each category offering different data transmission speeds and capabilities.

Cat5e and Cat6 are common choices for standard Ethernet connections in homes and businesses.

2. **Fiber Optic Cable**: Fiber optic cables use light signals to transmit data, offering high-speed, long-distance connectivity. They are commonly used in high-speed internet connections, data centers, and long-distance networking.
3. **Coaxial Cable**: Coaxial cables are often used for cable television (CATV) and broadband internet connections. They have a central conductor surrounded by insulating layers and a metallic shield.
4. **Twisted Pair Cable**: Twisted pair cables are used for telephone and data communication. They consist of pairs of insulated copper wires twisted together to reduce electromagnetic interference.
5. **USB Cable**: Universal Serial Bus (USB) cables are commonly used to connect various peripherals to computers. While not typically used for network connections, they serve a crucial role in connecting devices like printers, external hard drives, and more.
6. **Serial Cable**: Serial cables are used for connecting devices with serial interfaces, such as RS-232, to communicate and exchange data. They are common in industrial and embedded systems.
7. **Patch Cable**: A patch cable is a short network cable with connectors at both ends, typically used to connect devices in a local area network. They are often used to create patch cords or to connect devices to network switches or routers.

✓ **Cable connector**

A cable connector is a physical component that plays main role in connecting VoIP devices, such as VoIP phones or adapters, to the network infrastructure. These connectors are designed to establish a secure and reliable connection between the VoIP device and the network, ensuring the transmission of voice data over the internet or an IP-based network. Here are some key aspects of cable connectors in a VoIP system:

1. **Ethernet Connector**: The most common cable connector used in VoIP systems is the Ethernet connector. This connector typically follows the industry standard for networking cables and is used to link VoIP phones or adapters to the network through Ethernet cables. It usually has an RJ-45 interface, which is the standard connector for Ethernet cables.

2. **RJ-45 Connector**: An RJ-45 connector is a common type of cable connector used in Ethernet connections. It has eight pins and is used to attach Ethernet cables to network ports on VoIP devices and network equipment such as switches and routers.

3. **Power over Ethernet (PoE) Connector**: In some VoIP setups, power for VoIP phones is delivered through the same Ethernet cable used for data. PoE connectors allow both data and power to be transmitted through a single cable, simplifying installation and reducing the need for additional power sources.

4. **Connection to VoIP Devices**: The cable connector on a VoIP phone or adapter is typically located on the back or bottom of the device. Users plug an Ethernet cable with the appropriate connector into the VoIP device to establish the network connection.

5. **Cable Quality and Compatibility**: It's important to use high-quality and compatible cable connectors and Ethernet cables to ensure a reliable and stable network connection for VoIP devices. Properly terminated connectors and cables are essential for minimizing signal loss and ensuring good call quality.

6. **Structured Cabling**: In larger VoIP deployments, structured cabling systems are often used to organize and manage the network cables and connectors efficiently. This helps with scalability, maintenance, and overall network performance.

7. **Patch Cables**: Patch cables, which are typically shorter Ethernet cables with connectors on both ends, are commonly used to connect VoIP phones to network ports on switches or patch panels. They simplify the process of connecting devices in a structured cabling

- **VoIP System equipment**

  ✓ **Router**

A router is a network device that plays a pivotal role in managing and directing the flow of voice and data traffic over the internet or an IP network. Specifically, in the context of VoIP, a router performs the following functions:

1. **Packet Routing:** A VoIP router is responsible for routing voice and data packets between VoIP endpoints, such as IP phones, softphones, or VoIP gateways. It determines the best path for these packets to travel, ensuring that they reach their intended destinations efficiently.
2. **Quality of Service (QoS):** VoIP routers typically support QoS features that prioritize VoIP traffic over other types of data on the network. QoS mechanisms ensure that voice packets are given high priority, reducing the likelihood of delays, jitter, or packet loss, which can degrade call quality.
3. **NAT Traversal:** Network Address Translation (NAT) is commonly used in home and office networks. VoIP routers include NAT traversal capabilities to allow VoIP devices behind a NAT device to communicate with devices on the public internet. This is important for establishing VoIP calls across private networks.
4. **Firewall and Security:** VoIP routers often come with built-in firewalls to protect the VoIP network from security threats and unauthorized access. These firewalls can be configured to permit VoIP traffic while blocking other unwanted traffic.
5. **Bandwidth Management:** VoIP routers may offer bandwidth management and traffic shaping features. These features help allocate adequate bandwidth for VoIP calls, especially in situations where network resources are limited.
6. **Codec Support:** VoIP routers may support various audio codecs used to compress and decompress voice data. The choice of codec can impact call quality and the amount of bandwidth required for VoIP calls.
7. **VPN Support:** Some VoIP routers support Virtual Private Networks (VPNs) for encrypting VoIP traffic, adding an extra layer of security to VoIP communications.
8. **Configuration and Management:** VoIP routers have user-friendly interfaces for configuring VoIP settings, including SIP (Session Initiation Protocol) configurations and other VoIP-related parameters. Proper configuration is essential for a reliable VoIP service.

- ✓ PBX

A PBX, or Private Branch Exchange, is a telephone system used by organizations to manage internal and external phone calls. It serves as a central switching system that connects multiple telephones within an organization, enabling internal communication and access to external phone lines, typically through a service provider. There are some key features and functions of a PBX system:

1. **Call Routing:** A PBX routes incoming and outgoing calls to their intended destinations within the organization. This includes connecting calls between internal extensions and directing external calls to the appropriate recipient or department.
2. **Extension Dialing:** Users within the organization can dial short extension numbers to reach colleagues or departments, making it easy to communicate internally.
3. **Voicemail:** Many modern PBX systems include voicemail capabilities, allowing users to leave messages for one another when they are unavailable.
4. **Auto Attendant:** PBX systems often have auto-attendant features that provide a menu of options for callers, such as "Press 1 for Sales" or "Press 2 for Support." Callers can choose the appropriate option to be routed to the right department or extension.
5. **Call Transfer:** PBX systems allow users to transfer calls to other extensions or departments, ensuring that callers reach the right person or team.
6. **Conference Calling:** Many PBX systems support conference calling, allowing multiple parties to join a single call, which is useful for meetings and collaboration.

7. **Call Queues:** PBX systems can set up call queues for handling high call volumes. Calls are held in a queue and answered by available agents in the order they were received.
8. **Call Recording:** Some PBX systems have call recording capabilities for training, quality assurance, or legal compliance purposes.
9. **Reporting and Analytics:** PBX systems often provide reporting and analytics features that help organizations track call volumes, call duration, and other call-related metrics.
10. **Integration:** Modern PBX systems can integrate with other communication tools and software, such as customer relationship management (CRM) systems, to improve efficiency and customer service.
11. **Security:** PBX systems include security features to protect against unauthorized access, toll fraud, and eavesdropping.
12. **Remote Access:** Many PBX systems support remote access, allowing users to make and receive calls from external locations, which is especially valuable for remote and mobile workers.

✓ **PABX**

PABX is a business solution for companies that need many lines for in-house and outside calls. It allows companies to use a single access number that has several extensions. This is cheaper compared to using many landlines linked to the public network. A company that uses either PBX or PABX acts like a telephone exchange. PABX automates the switching tasks needed to connect calls between extensions. That’s the “A” in the acronym. In contrast, traditional PBX needs human switchboard operators to connect phone users.

In a PABX system, each device connected to the exchange has a designated extension number. These devices include desk phones, computer modems, and fax machines. A PABX system is often owned and administered by the company hosting it. Many call centers and large enterprises use PABX. Common features include auto attendant, call conferencing, call hold, and call transfer.

- ✓ **Phone**

A "phone" refers to a device or application used for making voice calls over the internet or an IP network. VoIP phones come in various forms and can be classified into several categories.

There are some common types of VoIP phones in a VoIP system:

1. **IP Phones (Hardware VoIP Phones):** These are physical devices that resemble traditional desk phones but are specifically designed to work with VoIP systems. IP phones connect to the internet or a local network and use the SIP (Session Initiation Protocol) or other VoIP protocols to make and receive calls. They typically have keypads, displays, and support for various VoIP features. IP phones can be corded or cordless (wireless).
2. **Softphones:** Softphones are software applications that run on computers, smartphones, or tablets. They allow users to make voice and video calls over the internet using a microphone and speakers or a headset. Softphones often include features such as call recording, conferencing, and instant messaging. Popular softphone applications include Skype, Zoom, and various other VoIP clients.
3. **Analog Telephone Adapters (ATAs):** ATAs are devices that enable traditional analog telephones to be used with VoIP systems. They convert analog voice signals from regular phones into digital data that can be transmitted over the internet. This allows organizations to transition gradually to VoIP while still using existing analog phones.
4. **Video Phones:** Some VoIP phones are equipped with video capabilities, enabling users to make video calls in addition to voice calls. These are particularly useful for video conferencing and remote collaboration.
5. **Web-Based VoIP Services:** Many web-based communication services, such as Google Voice and Microsoft Teams, provide integrated VoIP calling features. Users can make and receive calls directly through a web browser, without the need for any additional hardware or software.

6. **Mobile VoIP Apps:** There are mobile apps designed for smartphones that allow users to make VoIP calls over cellular data or Wi-Fi. Apps like WhatsApp, Viber, and Telegram provide voice and video calling features through the internet.
7. **Conference Phones:** These are specialized phones designed for conference rooms and larger meeting spaces. They often include features for multi-party calls, echo cancellation, and noise reduction to facilitate clear and effective conference calls.
8. **Cordless VoIP Phones:** Similar to traditional cordless phones, these devices are designed for mobility within a home or office while using a VoIP system. They connect to a base station, which is linked to the VoIP network.

✓ **Switch**

a switch plays a crucial role in managing the flow of voice and data traffic between different devices, such as phones, computers, and other endpoints. There are different types of switches used in VoIP systems:

1. **Ethernet Switch:** This is a fundamental component of any VoIP system. Ethernet switches manage the data traffic on your local network. They are responsible for ensuring that data packets are delivered to their intended destinations, including VoIP phones and other networked devices.
2. **VoIP Phone Switch:** This type of switch is sometimes called a "Voice Switch" or "IP PBX (Private Branch Exchange)." It's the central device that manages calls within an organization. VoIP phones connect to this switch, which then routes calls to their intended destinations. It can also provide features like call forwarding, voicemail, call recording, and more.

3. **Media Gateway:** A media gateway is used to convert analog voice signals to digital data packets and vice versa. It is necessary when connecting traditional analog phones or ISDN lines to a VoIP network. These gateways can also handle the conversion of different voice codecs used in VoIP.
4. **Session Border Controller (SBC):** An SBC is a device that manages and secures VoIP calls as they traverse between different networks or service providers. It helps protect the network from security threats, such as denial-of-service attacks and eavesdropping, and ensures that calls are properly established and terminated.
5. **Softswitch:** A softswitch is a software-based switch used to manage voice and data traffic in a VoIP network. It's more flexible and scalable compared to traditional hardware switches. Softswitches are used by service providers to route calls between different subscribers and handle various call management functions.
6. **Managed Switch:** In a corporate VoIP system, managed switches are often used. These switches offer more control over network traffic, allowing administrators to prioritize VoIP traffic to ensure call quality and minimize latency.
7. **PoE Switch:** Many VoIP phones require Power over Ethernet (PoE) for power. PoE switches provide both data connectivity and power to these phones through a single Ethernet cable, simplifying installation and reducing clutter.

- ✓ **Computer**

A computer in a VoIP (Voice over Internet Protocol) system can serve multiple purposes and play various roles within the VoIP ecosystem. There are some common ways in which computers are involved in VoIP:

1. **VoIP Softphones:** A VoIP softphone is a software application that allows a computer to function as a virtual telephone. Users can make and receive calls using a headset or microphone and speakers, often through a graphical user interface. These softphones can be standalone applications or integrated into unified communication software.

2. **VoIP Client Applications:** Apart from softphones, there are VoIP client applications used for communication and collaboration. These applications, such as Skype, Microsoft Teams, and Zoom, enable voice and video calls, instant messaging, and file sharing, making them a central part of VoIP communication.
3. **VoIP Servers:** Computers may be used to host VoIP servers, which are responsible for managing and routing VoIP calls. This includes PBX (Private Branch Exchange) servers, VoIP gateways, and other call processing components. These servers handle tasks like call routing, voicemail, call recording, and more.
4. **VoIP Conferencing:** Computers can be used to host and participate in VoIP conferencing systems. With software like Zoom or WebEx, multiple users can join audio and video conferences, enabling virtual meetings and collaboration.
5. **Network Management:** Computers are used to manage and monitor the VoIP network. Network administrators often use computers to configure and maintain the VoIP infrastructure, monitor call quality, and troubleshoot issues.
6. **VoIP Applications Development:** Developers may use computers to create custom VoIP applications, such as call center software, CRM integrations, or specialized communication tools. These applications can be used to enhance or extend VoIP capabilities.
7. **VoIP Testing and Quality Assurance:** Computers are used for VoIP quality testing and assurance. Specialized software can be used to simulate VoIP calls, measure call quality metrics, and diagnose network issues.
8. **VoIP Recording and Analytics:** In call centers and businesses, computers are used to record and analyze VoIP calls for quality control, compliance, and analytics. Software solutions can capture and store call recordings for later review and analysis.
9. **VoIP Security:** Computers play a role in VoIP security by hosting firewalls, intrusion detection systems, and security software to protect the VoIP network from threats such as unauthorized access, eavesdropping, and denial-of-service attacks.

Practical Activity 1.2.2: Conducting site survey.

Tasks:

Task 1: Read the given task.
You are a network engineer tasked with conducting a VoIP site survey at a client's location prior to installing a new VoIP system. The client has recently expanded their business and needs to upgrade their communication infrastructure.
Task 2: Trainees follow the instruction given by trainer.
Task 3: Trainees look what trainer demonstrates.
Task 4: Trainees do the task and present the work to the trainer and whole class.
Task 5: Ask clarification where necessary
Task 6: Read key **reading 1.2.2** and perform the task provided in application of learning 1.2.

Key readings:

Tools for Conducting VoIP Site Surveys

When conducting a VoIP site survey, you'll need a variety of tools to assess network infrastructure, bandwidth, and potential interference. Here are some essential tools:

Network Analysis Tools

- **Packet Analyzers:** These tools capture and analyze network traffic to identify issues like packet loss, latency, and jitter. Examples include Wireshark and tcpdump.
- **Network Scanners:** Used to map network topology, identify devices, and assess network performance. Examples include Nmap and Angry IP Scanner.
- **Bandwidth Testers:** Measure available bandwidth to ensure it meets the requirements of VoIP traffic. Tools like Speedtest.net and Ookla can be used.

VoIP-Specific Tools

- **VoIP Quality Monitoring Tools:** These tools monitor VoIP call quality metrics such as Mean Opinion Score (MOS), packet loss, and jitter. Examples include Asterisk, Kamailio, and FreeSWITCH.

- **VoIP Call Simulators:** Used to test VoIP call quality under different network conditions. They can simulate various scenarios like packet loss, latency, and jitter to assess the system's resilience.

Other Tools

- **RF Spectrum Analyzers:** Used to identify sources of interference, such as Wi-Fi signals or other radio frequency sources.
- **Sound Level Meters:** Measure background noise levels to ensure they do not interfere with VoIP calls.
- **Power Quality Analyzers:** Assess the stability and quality of the power supply, as fluctuations can affect VoIP equipment.

Additional Considerations:

- **Documentation:** Keep detailed records of your findings, including network diagrams, test results, and any identified issues.
- **Collaboration:** Work closely with the client's IT team or network administrator to gather necessary information and ensure a smooth survey process.
- **Follow-up:** After the survey, provide a comprehensive report detailing your findings and recommendations.

Points to Remember

- ✓ **Physical Site Survey:** This involves assessing the environment where the VoIP system will be installed, including factors like network infrastructure, power sources, and environmental conditions.

- ✓ **Tools:** Ethernet cable, punch-down tool, wire strippers, cable testers.
- ✓ **Materials:** RJ45 connectors, patch panels, wall plates, mounting hardware.
- ✓ **Equipment:** VoIP phones, PoE switches, VoIP gateways, VoIP PBX systems.

- ✓ When conducting a VoIP site survey, you'll need a variety of tools to assess network infrastructure, bandwidth, and potential interference such as Packet Analysers, Network Scanners, Bandwidth Testers, VoIP Call Simulators, RF Spectrum Analysers, Sound Level Meters and Power Quality Analysers.

Application of learning 1.2.

You are a network engineer tasked with conducting a VoIP site survey at a client's location prior to installing a new VoIP system. The client has recently expanded their business and needs to upgrade their communication infrastructure.

Indicative content 1.3: Design of VoIP System Diagram

Duration: 4 hrs

Theoretical Activity 1.3.1: Description of Physical diagram

Tasks:

Task 1: Answer the following questions related to the description of physical diagram.

i. What is physical diagram?

ii. What is a physical network diagram?

iii. Mention types of physical topology

iv. How do you draw a physical diagram?

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to the whole class

Task 4: For more clarification, ask questions where necessary.

Task 5: Read the key readings 1.3.1.

Key readings:

- **Description of Physical diagram**

✓ **What Is a Physical Network Diagram?**

As we have often heard that visuals make an impression on the mind rather than simple theory or text. If we talk about the IT infrastructure, which is flooded with a range of devices, Network diagrams act as a one-stop solution in the scenario. It is a schematic that depicts the connections among all the devices in a network, such as servers, routers, and firewalls. It maps out the network architecture and visually explains the different components and connections in a network through symbols, shapes, or icons. An exemplary method to share the layout of any network, it makes the process easier to understand for users.

Depending on the requirements, a network diagram can either be very simple or very typical. They are both categorized into two types - physical and logical.

A physical network diagram illustrates the interconnection of the devices in the network with wires and cables while a logical diagram Illustrates the way information flows through a network. To be precise, the physical network diagram reveals the network topology with all the physical aspects, such as ports, cables, racks, servers, specific models, etc. usually used by IT professionals, they are used to visualize the communication scheme of the network arrangement in residences or offices.

- ✓ **Why Use a Physical Network Diagram?**

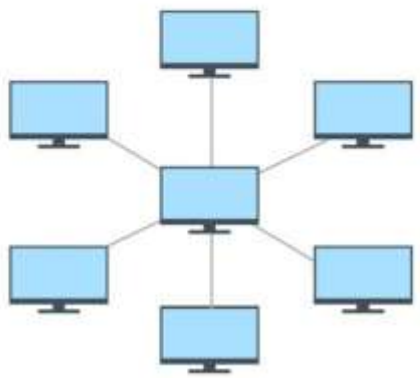
The main purpose of physical network diagrams is to provide insights about the different physical components in a network. Another purpose of a physical network diagram is to visualize how every component in the network works to assist with troubleshooting. A clear picture of the elements working together will pinpoint what needs to be remedied when something in the network is not working correctly.

A physical diagram is generally used to visualize the interconnection of network devices and other physical components. It displays the topology and the existence of physical links between devices. Mainly used by IT staff, a physical diagram visually documents the physical connections. They are used to troubleshoot network problems, identify security threats and weak spots, and organize changes in the network.

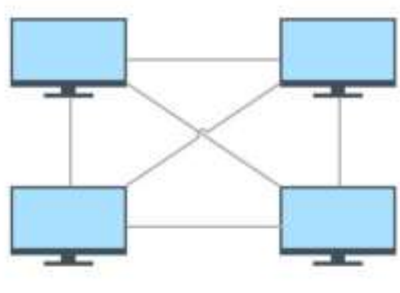
- ✓ **Types of the Physical Topology**

The physical topology is the interconnected structure of a local area network (LAN). All the physical devices, when connected to the network through several types of cables, together constitute in making the physical topology.

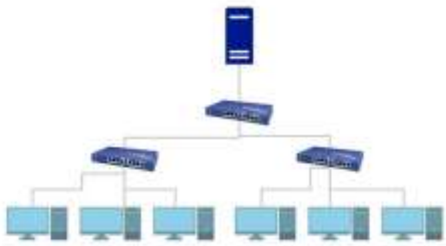
- **Star Topology:** This topology is employed with a single access point or a switch that acts as the centre of the network. All nodes are directed from this singular point.



- **Mesh Topology: Mesh topology** is a network of wires and cables that connects the computers and the network devices. All nodes are connected in a mesh topology.



- **Tree (Extended Star) Topology:** this typology is employed with multiple access points connected linearly. Then odes are directed from their respective access points.



- ✓ **Draw A Physical Network Diagram in EdrawMax?**

Step 1: Establish a hierarchy of levels, look out for the components you need to include for the given network, such as servers, routers, hubs, printers, and others.
Step 2: After mapping out the components on the diagram, the next step requires you to connect them. Add the relevant symbols. EdrawMax loads the shapes, tools, etc. beginning with inserting the basic components such as computers, servers, firewalls, etc.
Step 3: Label the symbols to depict the components and connections between your networks. Add to the legend for reference.
Step 4: To represent different types of connections, use different types of lines and different levels of thickness. Use lines and arrows to depict the connections.

**Practical Activity 1.3.2: Designing Physical diagram.**

Tasks:

Task 1: Referring to key reading (1.3.2). Perform the given task. The task should be done individually.
As learner who studied to physical diagram design Physical VoIP system diagram that comprise the PBX, ROUTER, Endpoint and iPhone switch in VoIP system.
Task 2: List out procedures to be used to open the drawing software tools.
Task 3: Follow the instruction given by trainer
Task 4: Referring to procedures provided by trainer, Perform the given tasks.
Task 5: Present your work to the trainer and whole class.
Task 6: Ask clarification where necessary
Task 7: Read key reading 1.3.2 and perform the task provided in application of learning 1.3.

Key readings: Designing Physical diagram.

A VoIP system physical topology diagram illustrates the physical connections between the various components of the system. Here are the steps involved in designing such a diagram:

1. Identify Components:

- **PBX (Private Branch Exchange):** The central switching system that handles calls within the network.
- **Router:** Connects the PBX to the external network (e.g., internet).
- **Endpoint:** The devices used to make and receive calls (e.g., IP phones, softphones).
- **iPhone:** A specific type of endpoint used for mobile VoIP.

2. Determine Network Structure:

- **Centralized:** All components are connected directly to the PBX.
- **Distributed:** Components are distributed across multiple locations, connected via a network.
- **Hybrid:** A combination of centralized and distributed structures.

3. Draw the Basic Diagram:

- **Place Components:** Position the PBX, router, and endpoints on the diagram.
- **Connect Components:** Use lines to represent physical connections.
- **Label Components:** Label each component for clarity.

4. Consider Network Topology:

- **Star Topology:** All devices are connected to a central hub or switch.
- **Mesh Topology:** Devices are connected to multiple other devices, creating a redundant network.
- **Ring Topology:** Devices are connected in a circular fashion.
- **Hybrid Topology:** A combination of different topologies.

5. Include Additional Components:

- **Gateway:** If connecting to the PSTN (Public Switched Telephone Network), include a gateway.
- **Firewall:** To protect the network from external threats.
- **Media Server:** For handling media processing tasks.
- **Presence Server:** For managing user presence information.

6. Specify Connectivity:

- **Wired or Wireless:** Determine if connections will be wired (using Ethernet cables) or wireless (using Wi-Fi).
- **Network Protocols:** Specify the network protocols used (e.g., TCP/IP, SIP).

7. Consider Scalability:

- **Redundancy:** Include redundant components (e.g., backup power supplies, redundant servers) to ensure high availability.
- **Modular Design:** Design the system to be modular, allowing for easy expansion and changes.

8. Verify Accuracy:

- **Review Diagram:** Ensure the diagram accurately represents the physical connections and components.
- **Consult Experts:** If necessary, consult network engineers or VoIP experts for verification.

Example Diagram:

VoIP system physical topology diagram with PBX, router, IP phones, and iPhone

Note: The specific components and connections will vary depending on the size and complexity of the VoIP system, as well as the desired features and functionalities.

Theoretical Activity 1.3.3: Description of Logical diagram and Use of design tools

Tasks:

Task 1: Answer the following questions related to the description of Logical diagram and use of design tools

i. What are logical network diagrams?
ii. Outline the logical network design tools.
iii. Give at least three the designing software tools.

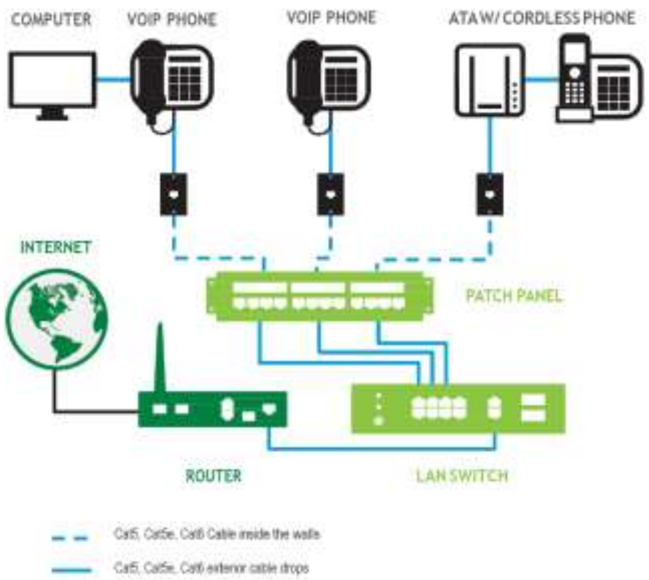
Task 2: Provide the answer for the asked questions and write them on papers.
Task 3: Present the findings/answers to the whole class
Task 4: In addition, ask questions where necessary.
Task 5: For more clarification, read the key readings 1.3.3.

Key readings: Description of Logical diagram and Use of design tools

- **Description of Logical diagram**
  - ✓ **What are logical network diagrams?**

A logical network diagram illustrates the flow of information through a network and shows how devices communicate with each other. It typically includes elements like subnets, network objects and devices, routing protocols and domains, voice gateways, traffic flow and network segments.

In logical network diagrams, there are pivots for small, medium and large networks, where network diagram templates can be helpful.



 **Practical Activity 1.3.4: Designing logical diagram and Using design tools.**

. Tasks:

Task 1: Referring to key reading (1.3.4) you are requested to perform the given task. The task should be done individually.

As network technician Using packet tracer as logical diagram tool design logical VoIP system by referring to key reading 1.3.4.

Task 2: List out procedures to be used to perform the given tasks.

Task 3: Follow the given instructions

Task 4: Referring to procedures provided on task (1.3.4), Perform the given tasks.

Task 5: Present your work to the trainer and whole class.

Task 6: Ask clarification where necessary

Task 7: Read key reading 1.3.4 and perform the task provided in application of learning 1.3.

 **Key readings: Designing logical diagram and Using design tools.**

- **Use of design tools**
  - ✓ **List of VoIP network design tools**
    - **Packet tracer**

Packet Tracer is a network simulation and visualization tool developed by Cisco Systems. It enables users, especially students and network professionals, to design, configure, and experiment with network topologies in a simulated environment, mimicking real-world networking scenarios. Packet Tracer facilitates hands-on learning and testing of Cisco devices and configurations, making it an essential resource for networking education and training.
It allows users to create, connect, and troubleshoot network elements like routers, switches, and end devices, helping them gain practical experience in network design and troubleshooting without the need for physical hardware.

- **eDraw max**

Edraw Max is a versatile and comprehensive diagramming software tool that offers a wide range of templates and tools for creating various types of diagrams, including flowcharts, org charts, network diagrams, and more. It's designed for professionals, students, and anyone seeking to visualize ideas and data effectively. Edraw Max provides an intuitive interface, extensive libraries of shapes and symbols, and collaborative features, making it a valuable solution for diagram creation, project planning, and data visualization across different fields and industries.

- **GNS3**

GNS3, short for "Graphical Network Simulator-3," is an open-source network emulation platform widely used by network engineers, students, and IT professionals for designing, configuring, and testing complex network topologies. GNS3 allows users to virtually emulate real hardware devices, such as routers, switches, and firewalls, and run actual networking operating systems. It offers a highly flexible and interactive environment for creating, modifying, and troubleshooting network configurations, making it an invaluable tool for learning, development, and testing of network setups and scenarios.

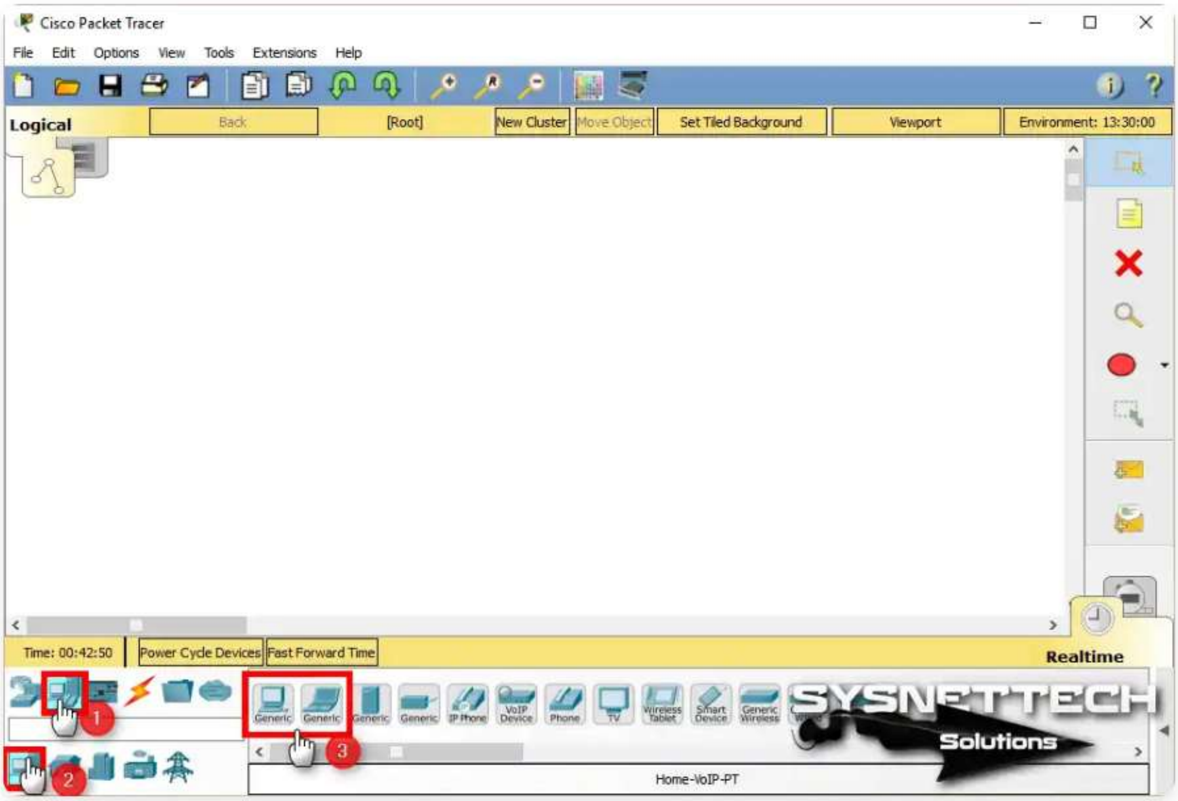
✓ **How to use Packet trace, eDraw max and GNS3.**

- **Packet tracer**

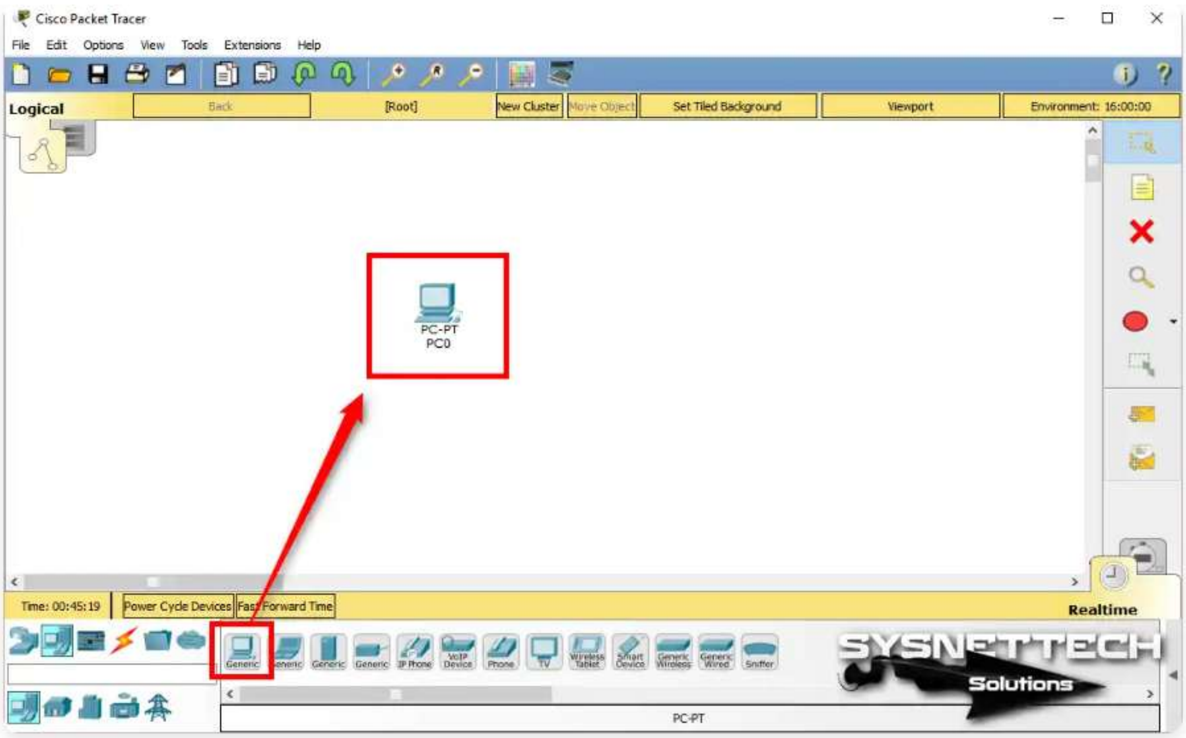
Using **Packet Tracer**, you can create detailed projects by creating network topologies or creating a virtual environment for your existing network with this software. Or, if you are preparing for Cisco exams, you can work with this simulator software on all topics covered by exam content, such as routing protocols and VLANs.

Step 1: How to Add a PC

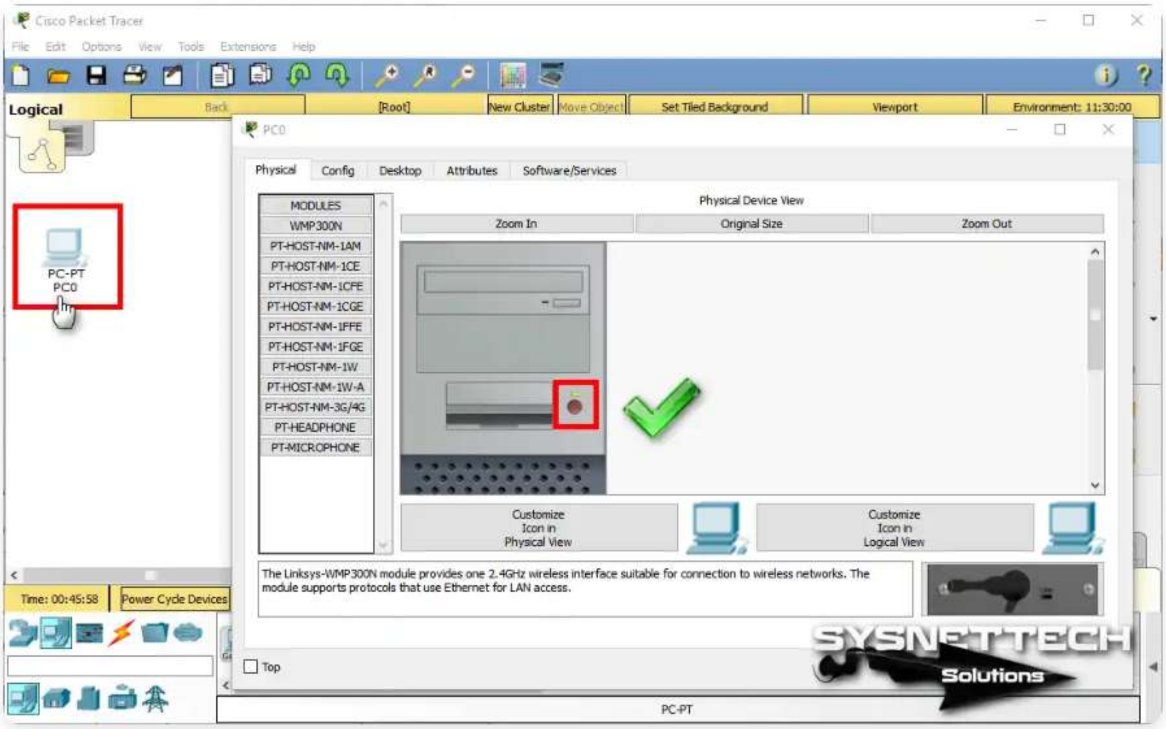
The most straightforward and essential task in Packet Tracer is to add a PC to the workspace. To add a PC, click the section below.



Drag and drop the device with the desktop view into the workspace in the right section.

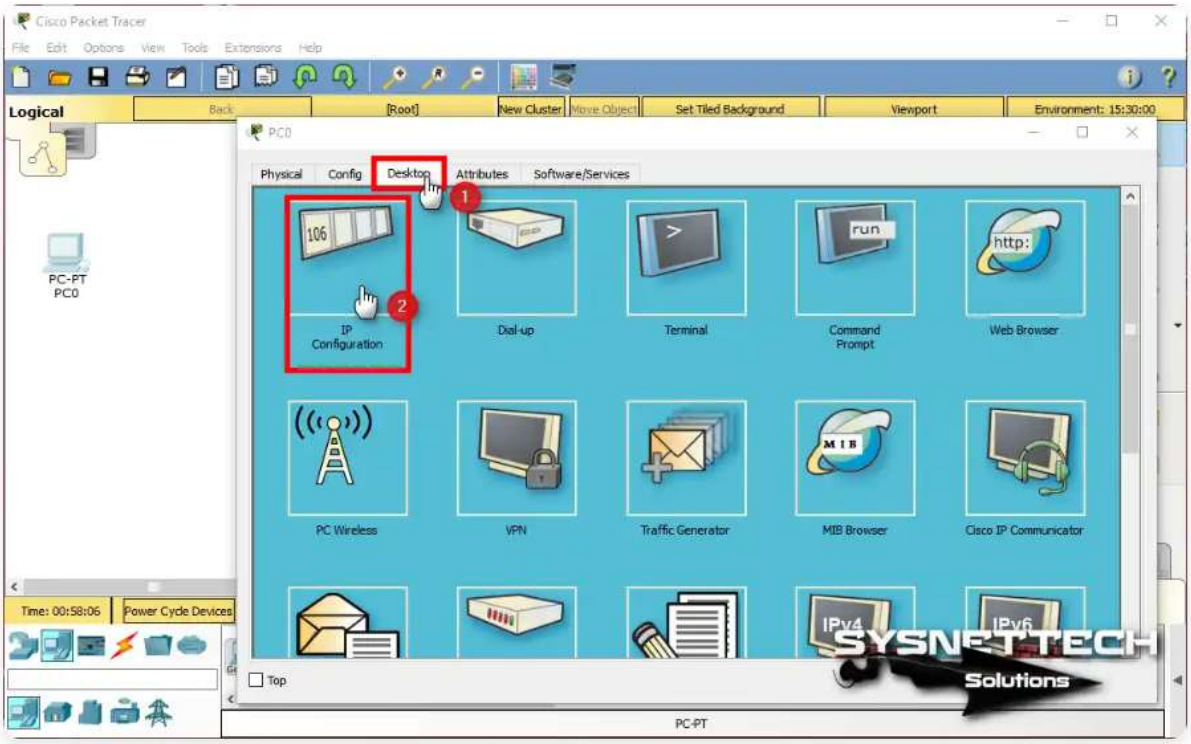


After adding a PC, you can open the corresponding hardware settings window by double-clicking on it. In this window, you can add modules to the PC from the area on the left, as in the image below. You can also turn the PC on or off from the section shown in the illustration.



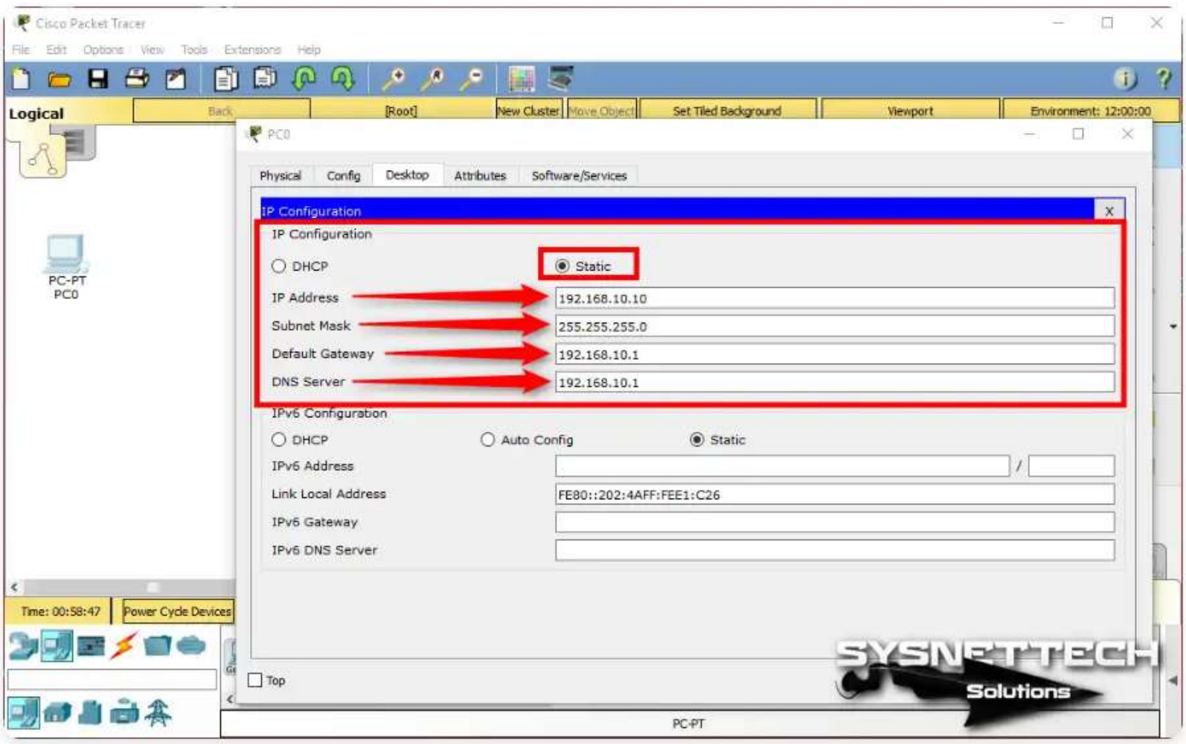
Step 2: How to Assign IP Addresses to Computers

You must assign IP addresses, subnet masks, and default gateway addresses to the PCs you add. For example, click to configure the TCP/IP settings of PC0, click the Desktop tab in the window that opens, and then click IP Configuration.

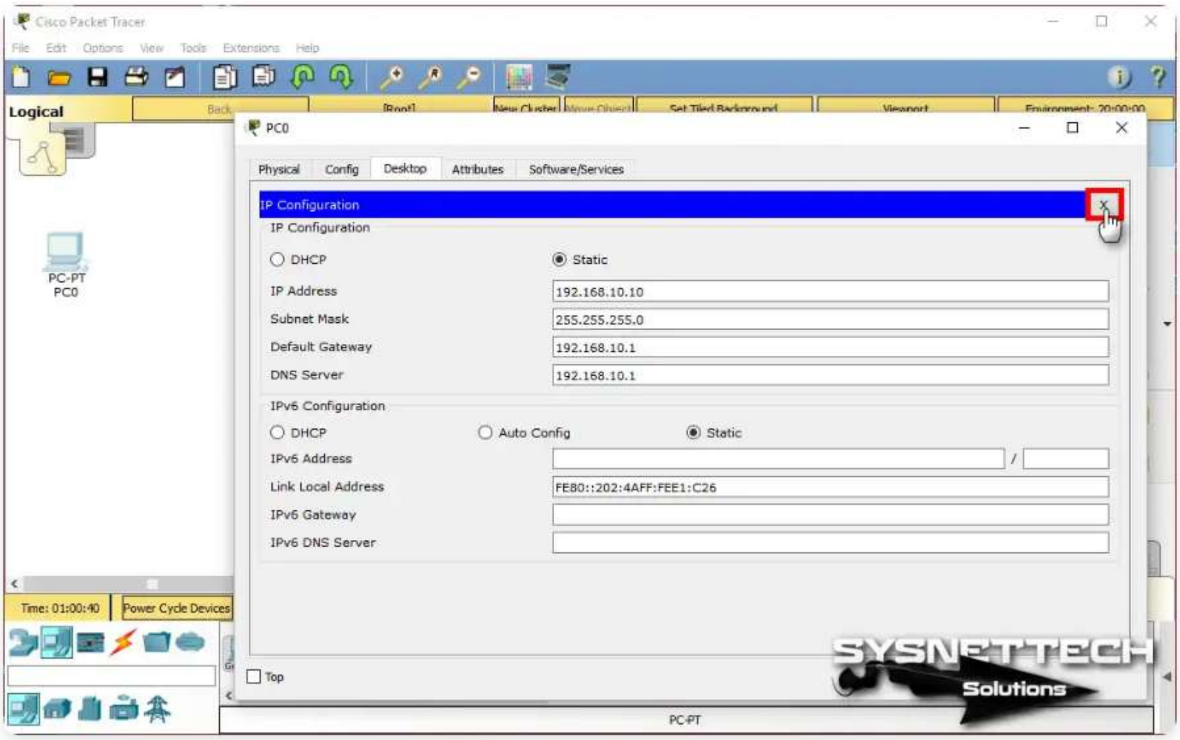


In the IP Configuration window, if you have a DHCP Server in your network topology, enable the DHCP option. If you do not have a server, you must select Static to manually assign an IP address and configure the IP settings as follows.

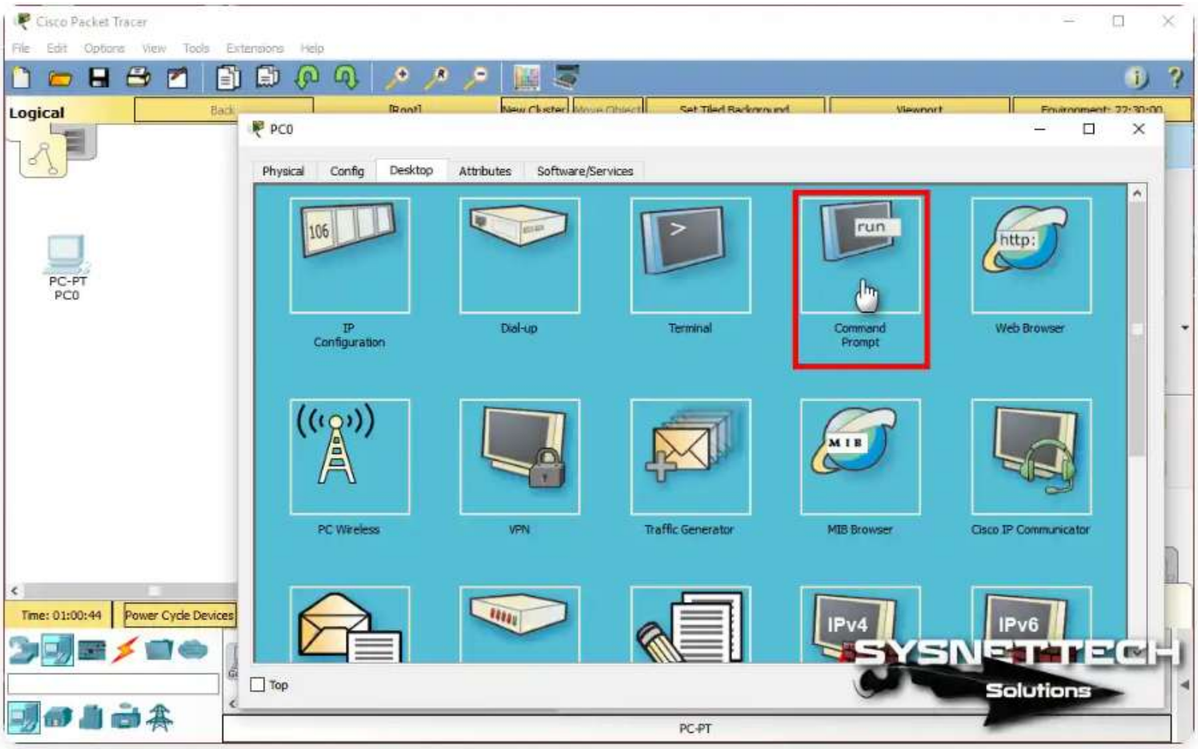
Configure the IP address settings for PC0 as follows.



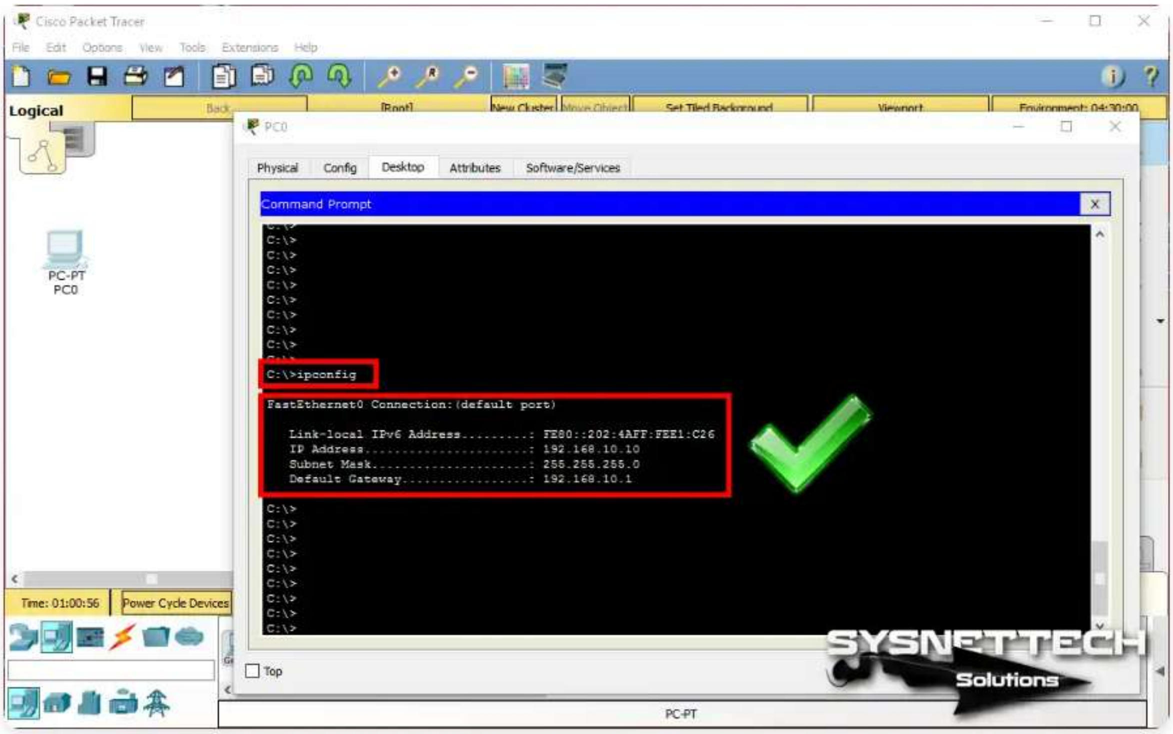
You do not need to save your settings because as soon as you click the X button, your settings will be saved.



Now, from the PC0 properties, click on Command Prompt.

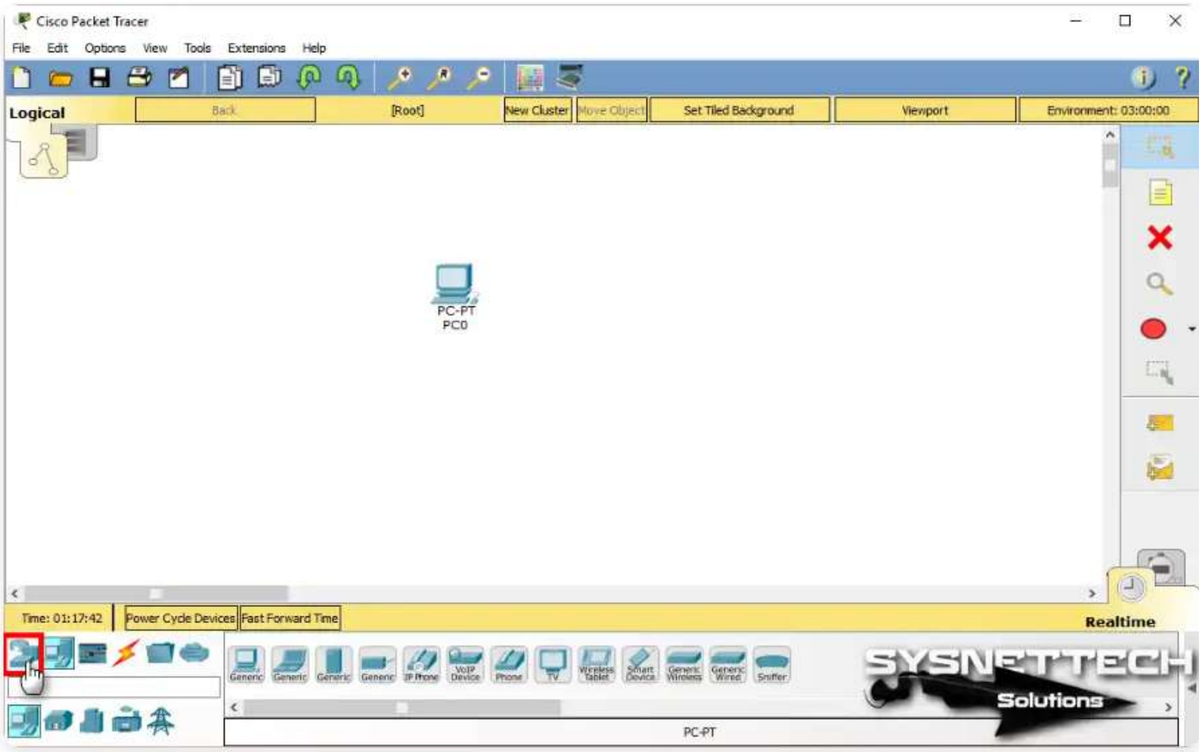


When you execute the ipconfig command on the CMD, you can see the IP settings of PC0.

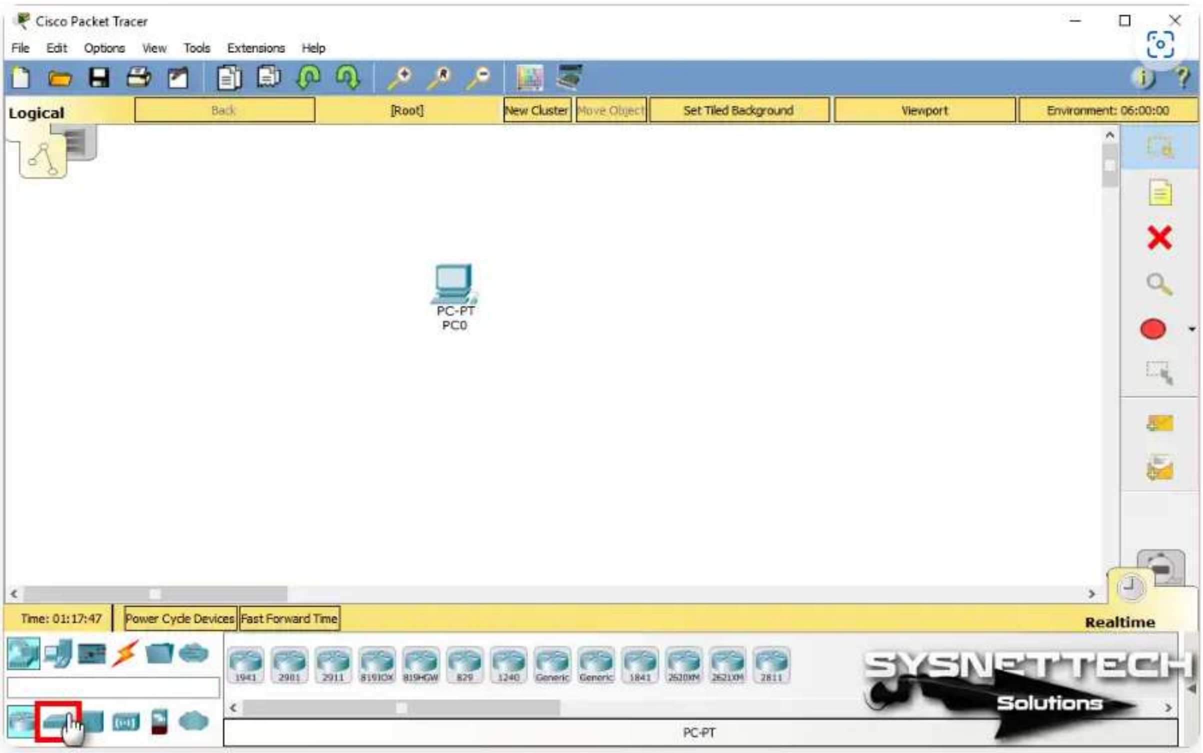


Step 3: How to Add a Cisco Switch

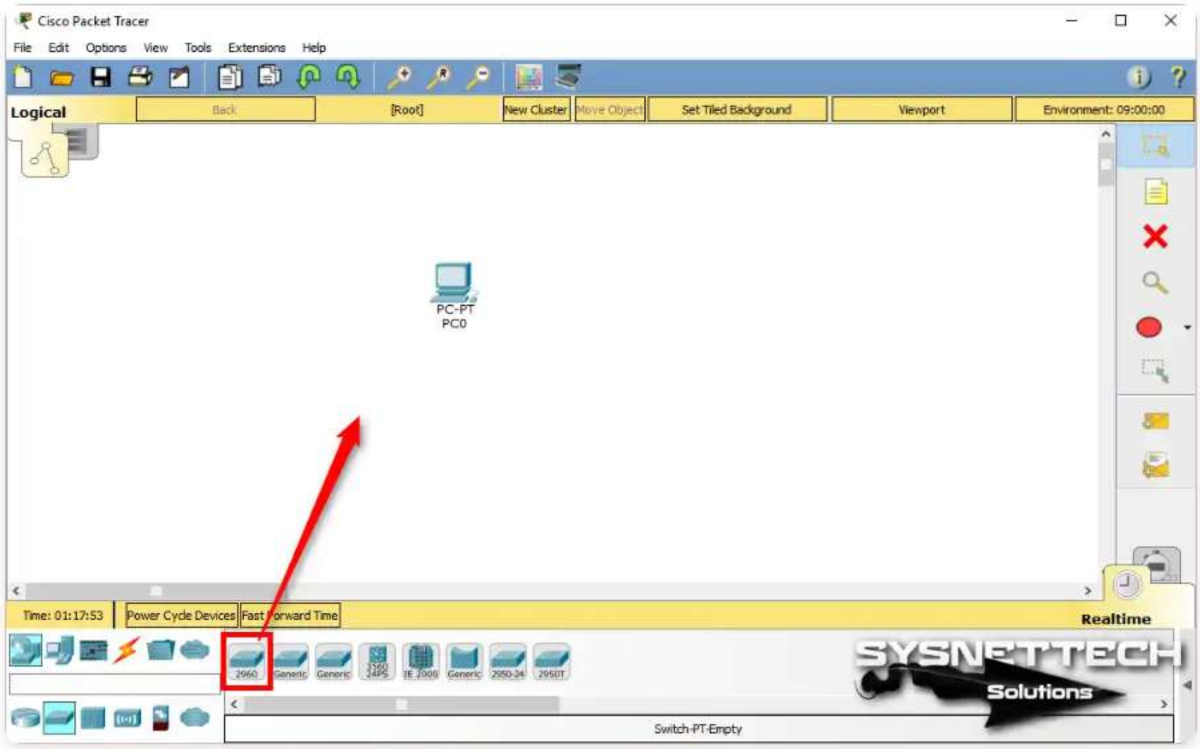
To add a Cisco Switch to the workspace, click (Network Devices) and click Switches from the options listed.



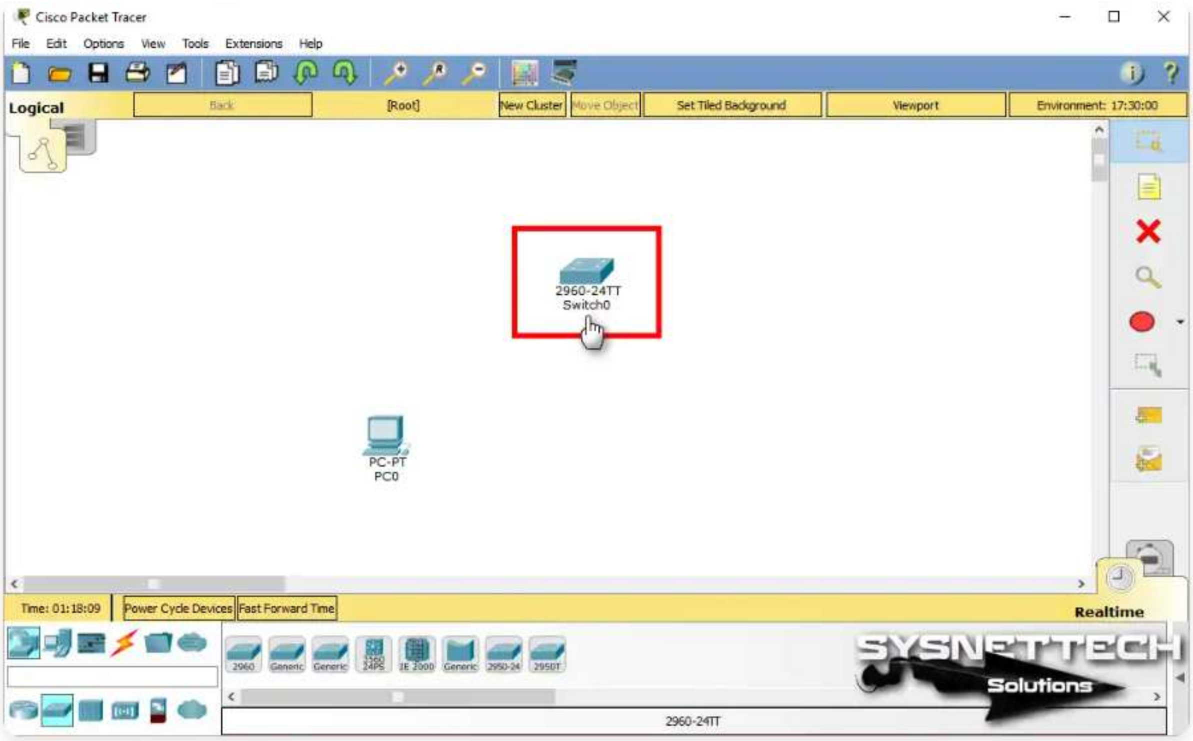
Under Network Devices, click Switches.



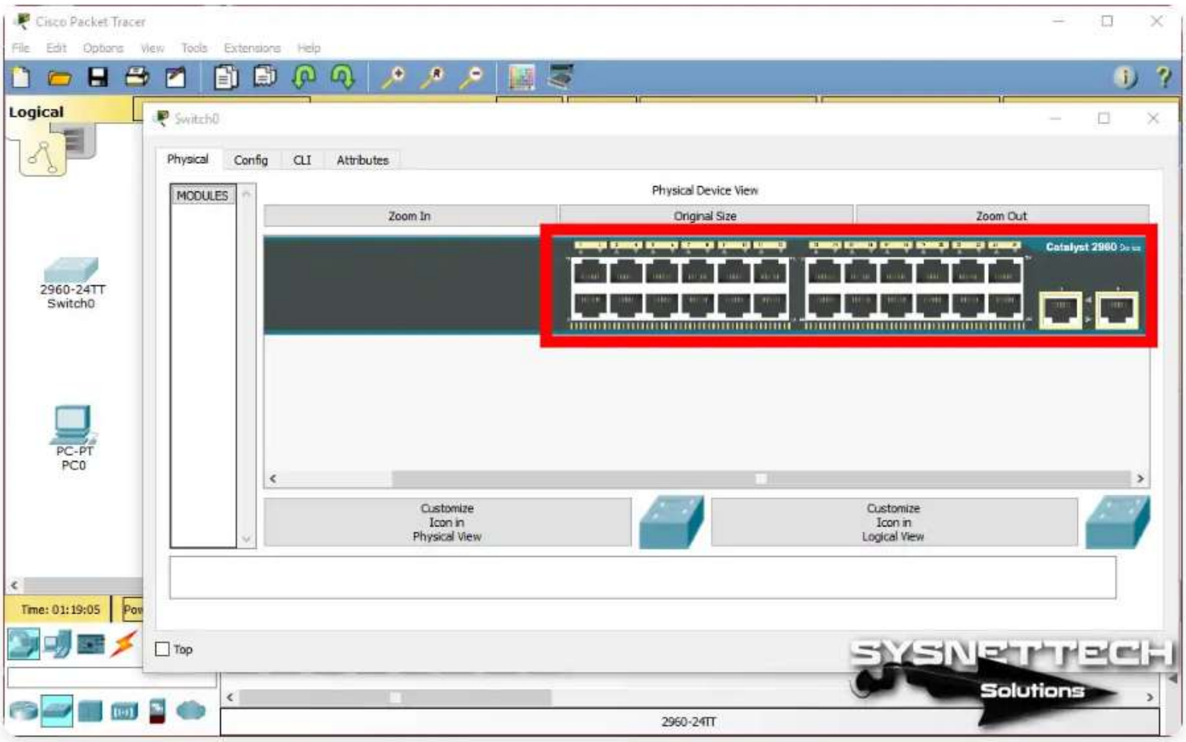
After clicking the switches, the device models will be listed in the middle pane. Drag and drop the relevant Switch model from the workspace to your network.



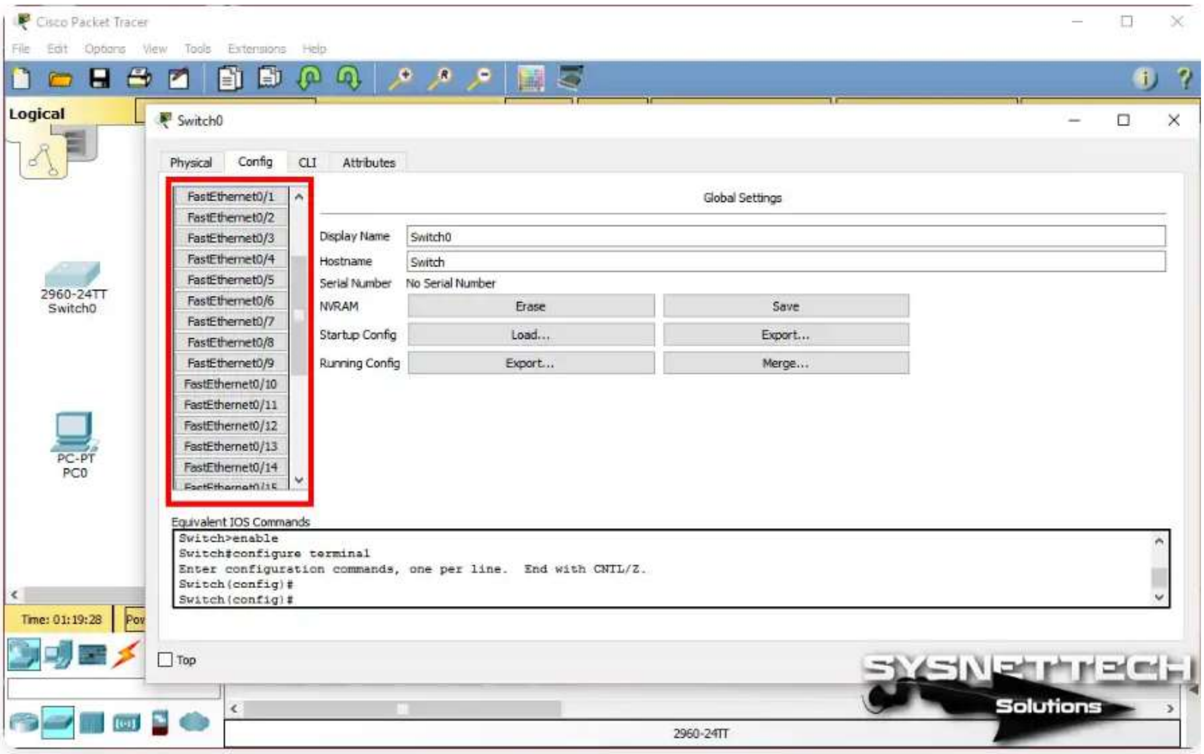
Once you have added the key to the workspace, click once to open its properties.



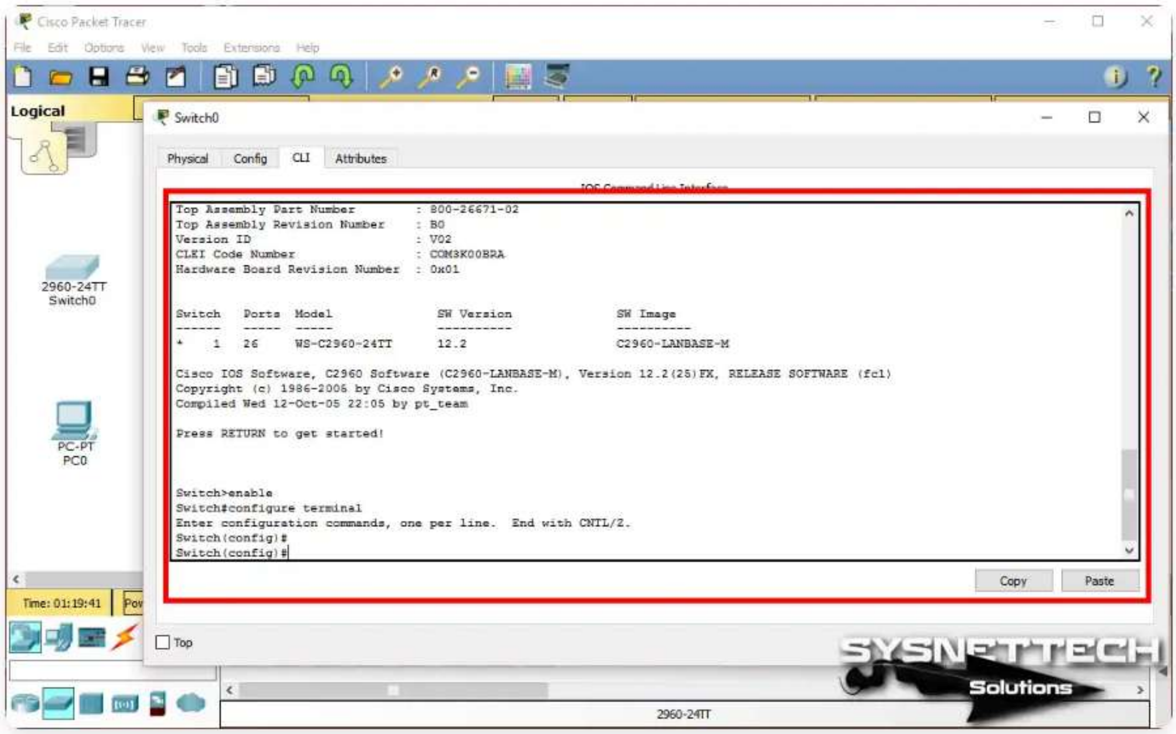
You can examine the physical appearance of the Switch in the Properties window.



After clicking the Config in the Switch0 properties, you will see some settings related to the device. From here, you can change the name and hostname of the Switch in the workspace. In addition, you can delete, save, and export Startup-Config and Running-Config files.



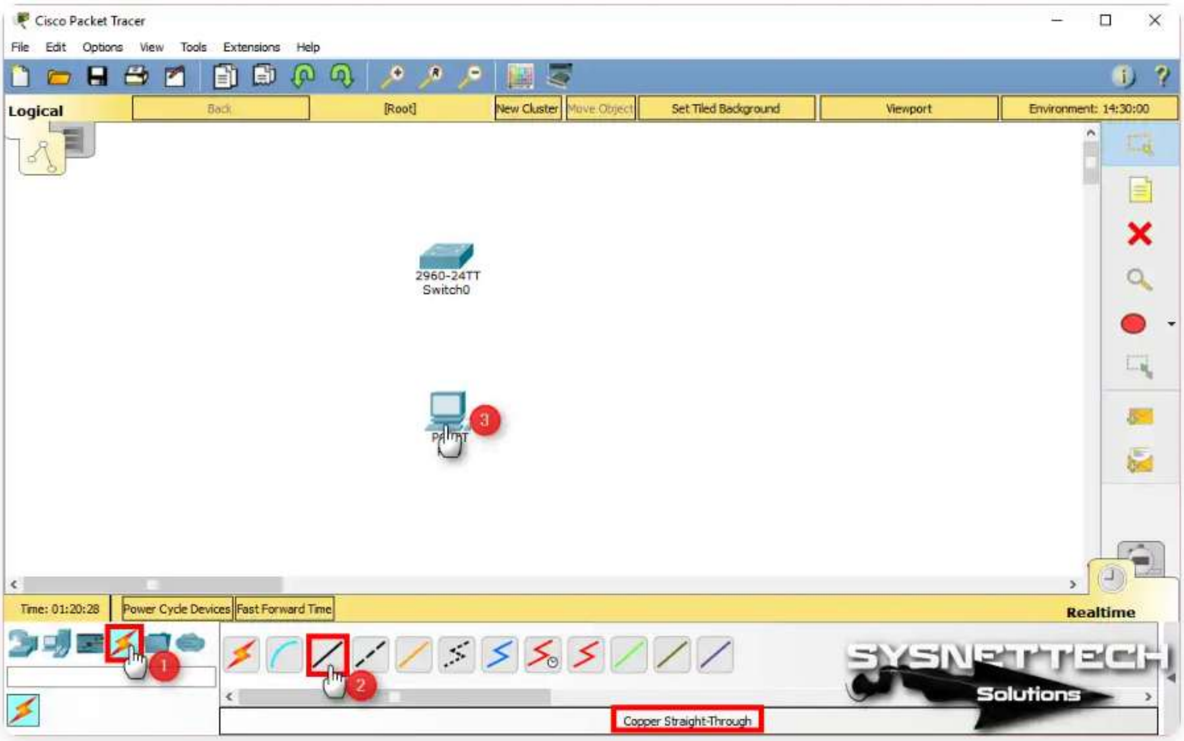
If you want to configure the Switch using commands, you can use the **IOS Command Interface.**



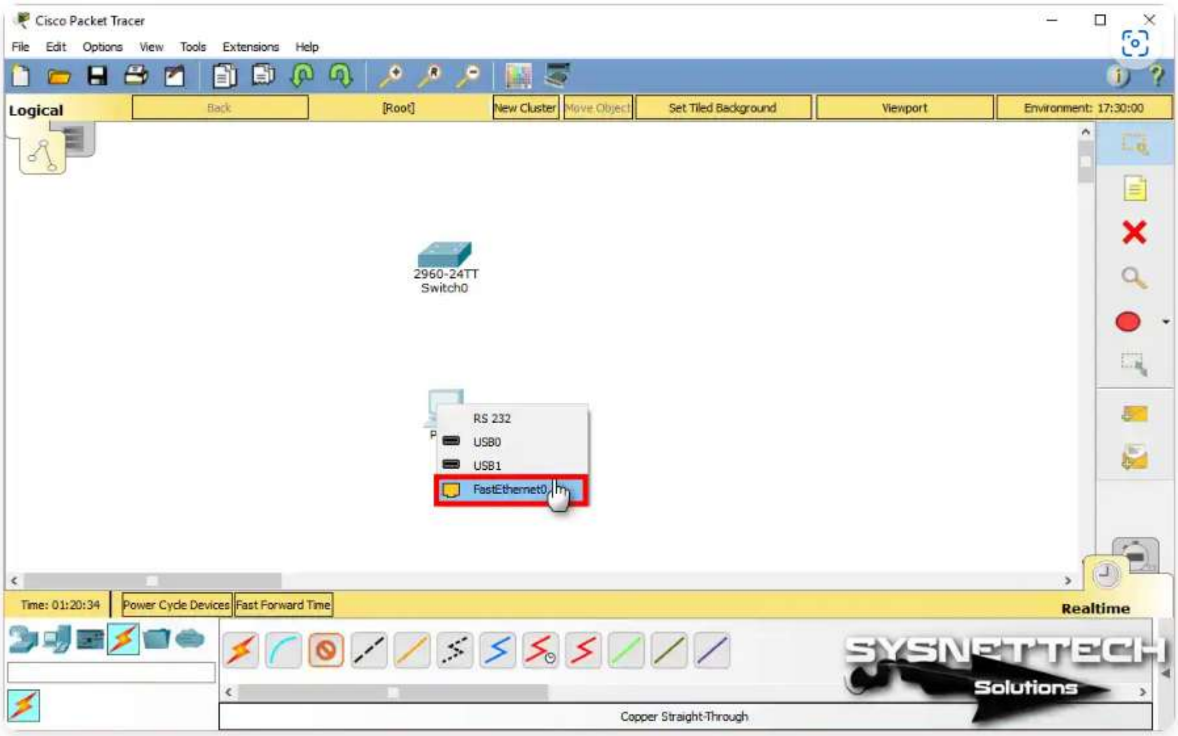
Step 4: How to Wire Network Devices

After adding a PC and Switch to the workspace, you must connect these devices with a network cable. To cabling network devices in Packet Tracer, list the cable types that the program supports by clicking 1 and then 2 as shown in the image below.

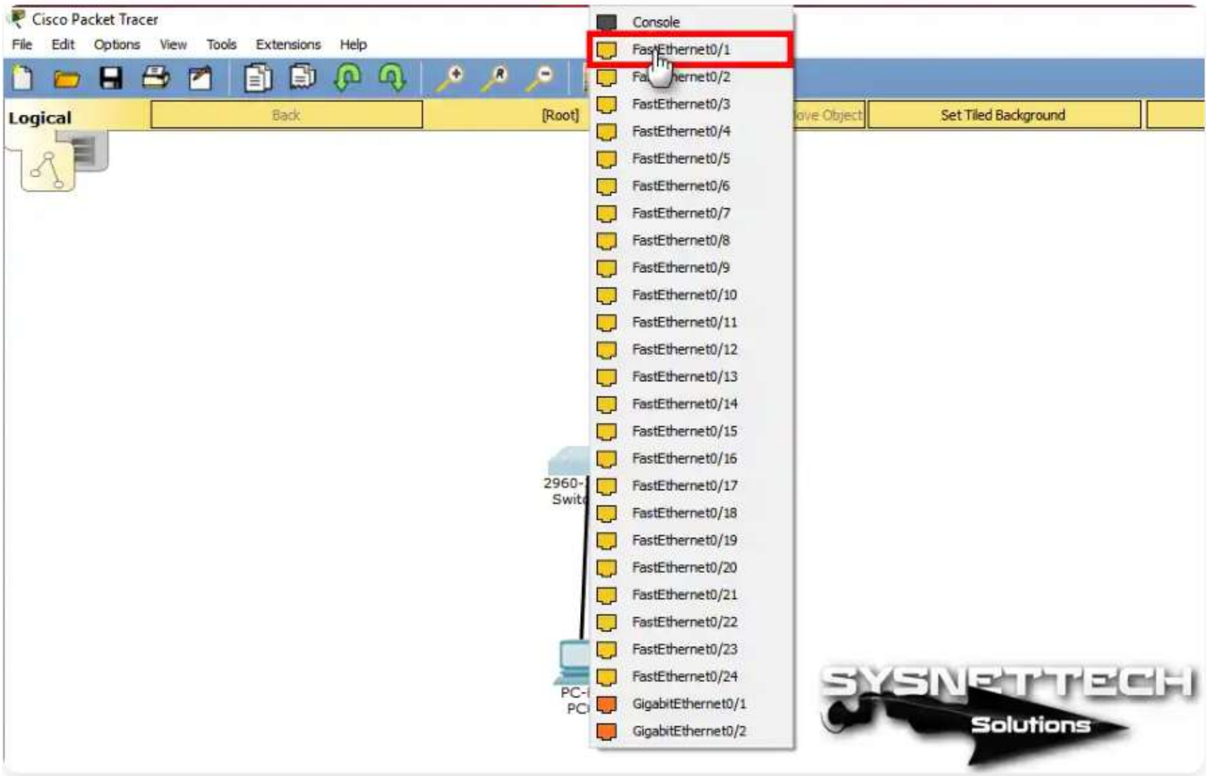
From the network cable types, select the straight-through cable type and click PC0.



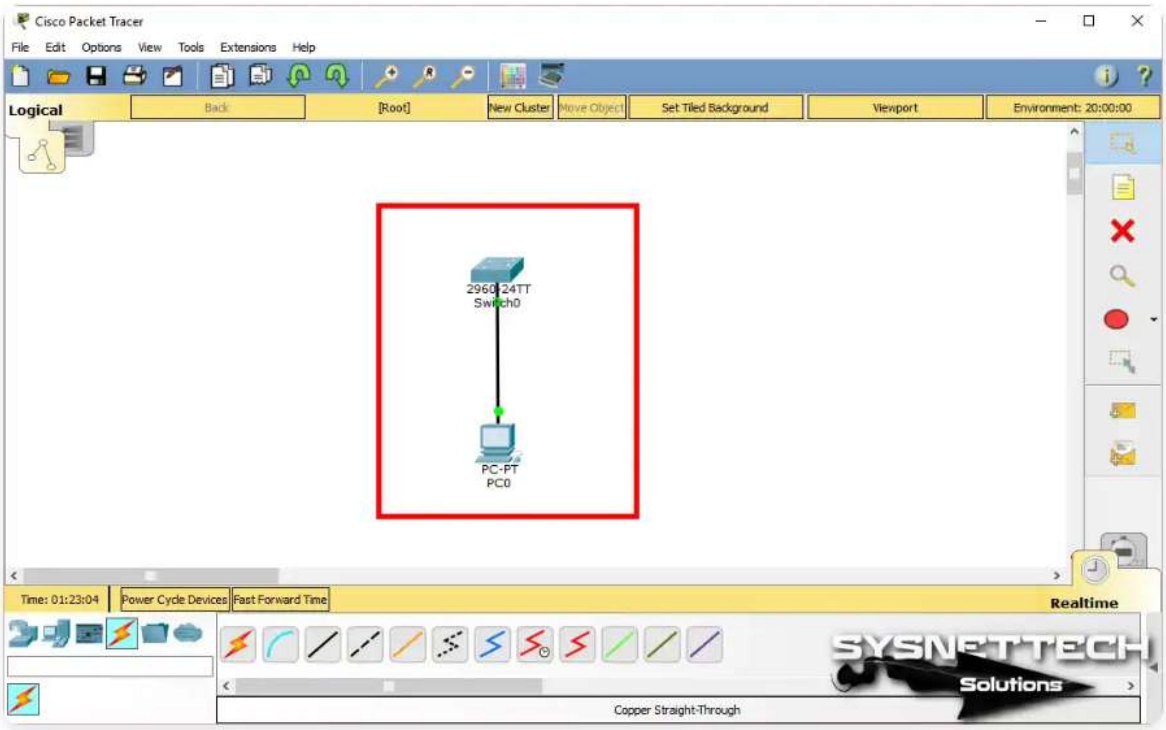
Select the FastEthernet0 network card on PC0.



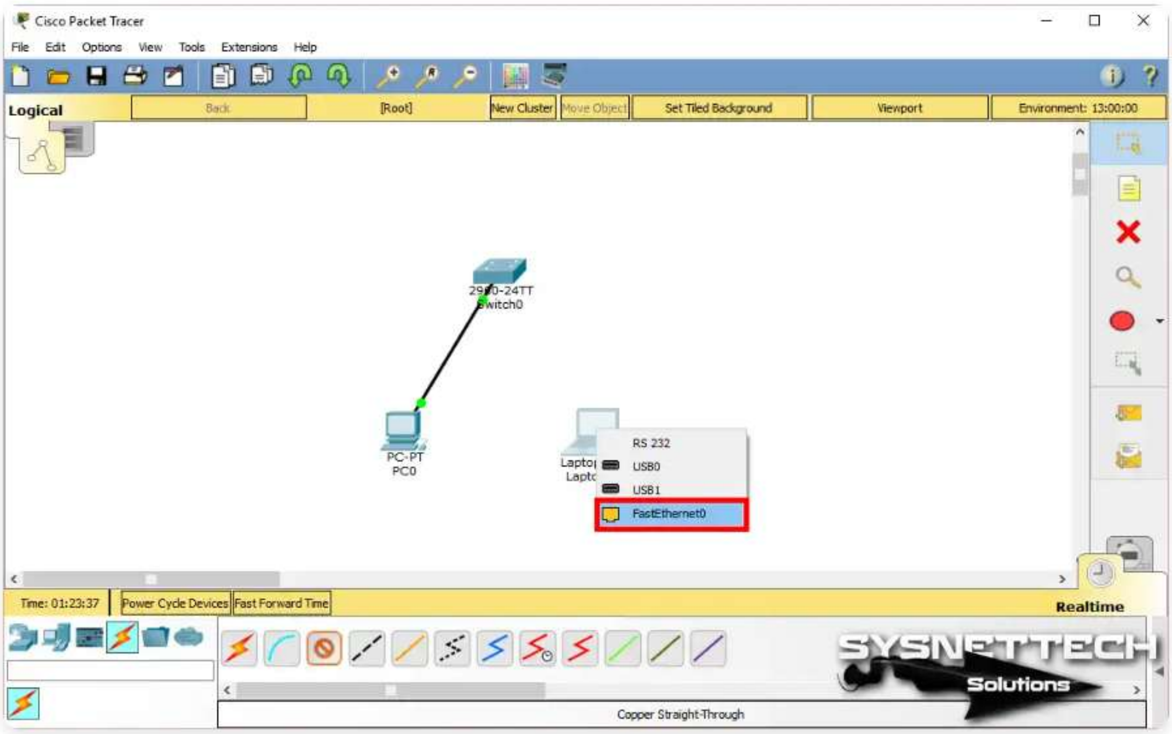
Now click Switch0 to connect the PC to the Switch and select any of the free ports listed.



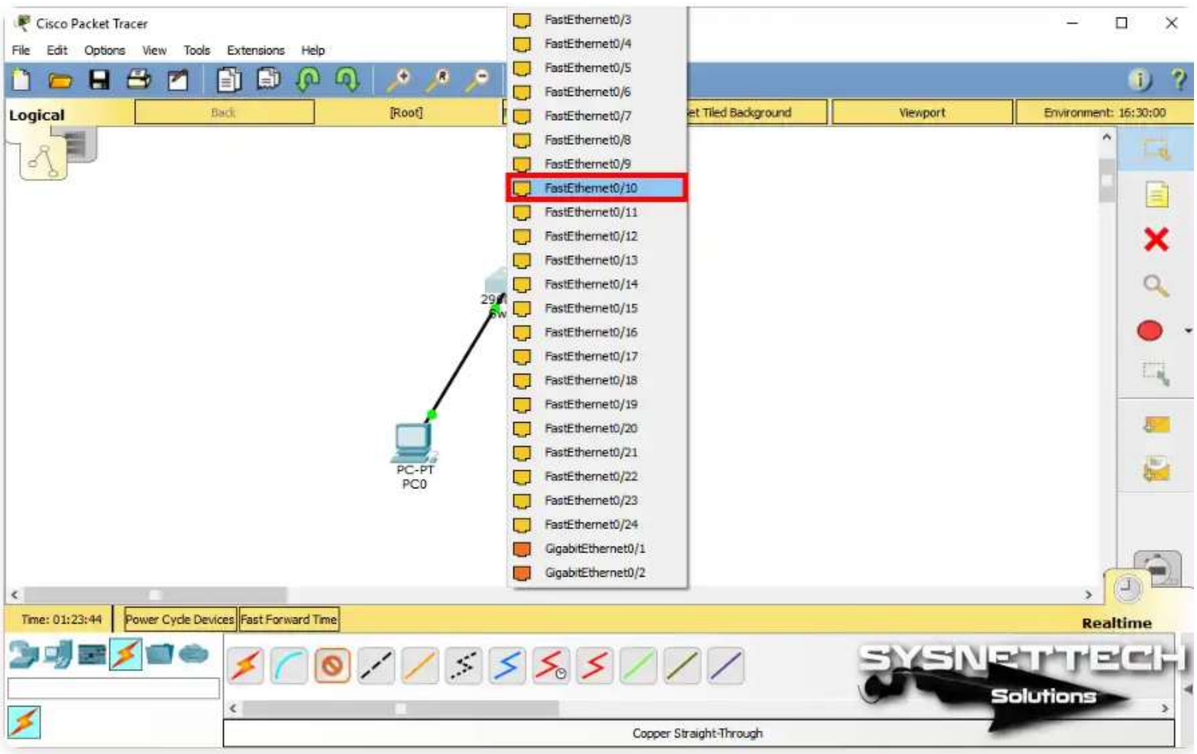
After connecting the network cable, the port status color between the PC and the Switch will be green, as in the following image. If the port is green, it means that the cable type is correct and working.



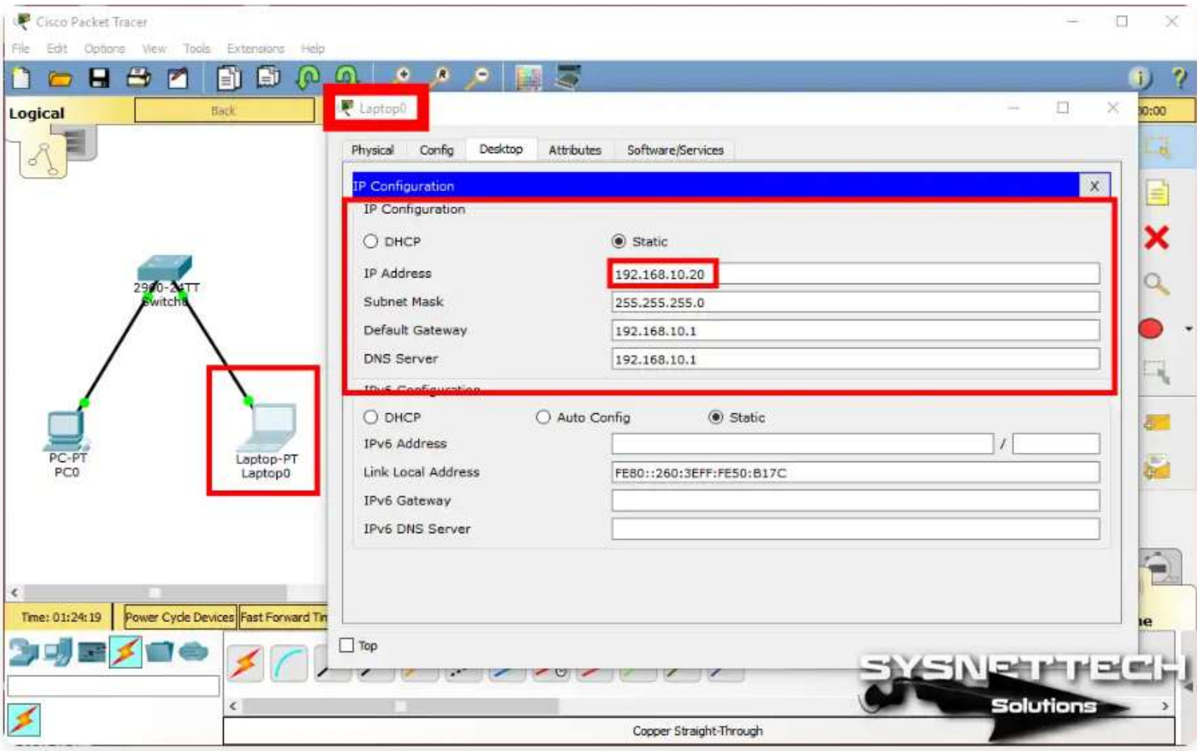
Add one Laptop to the network topology and click the wiring option in the same way to connect it to the Switch. Then, plug the network cable into the FastEthernet0 interface of the Laptop.



Also, plug the other end of the cable into an empty port on the Switch.

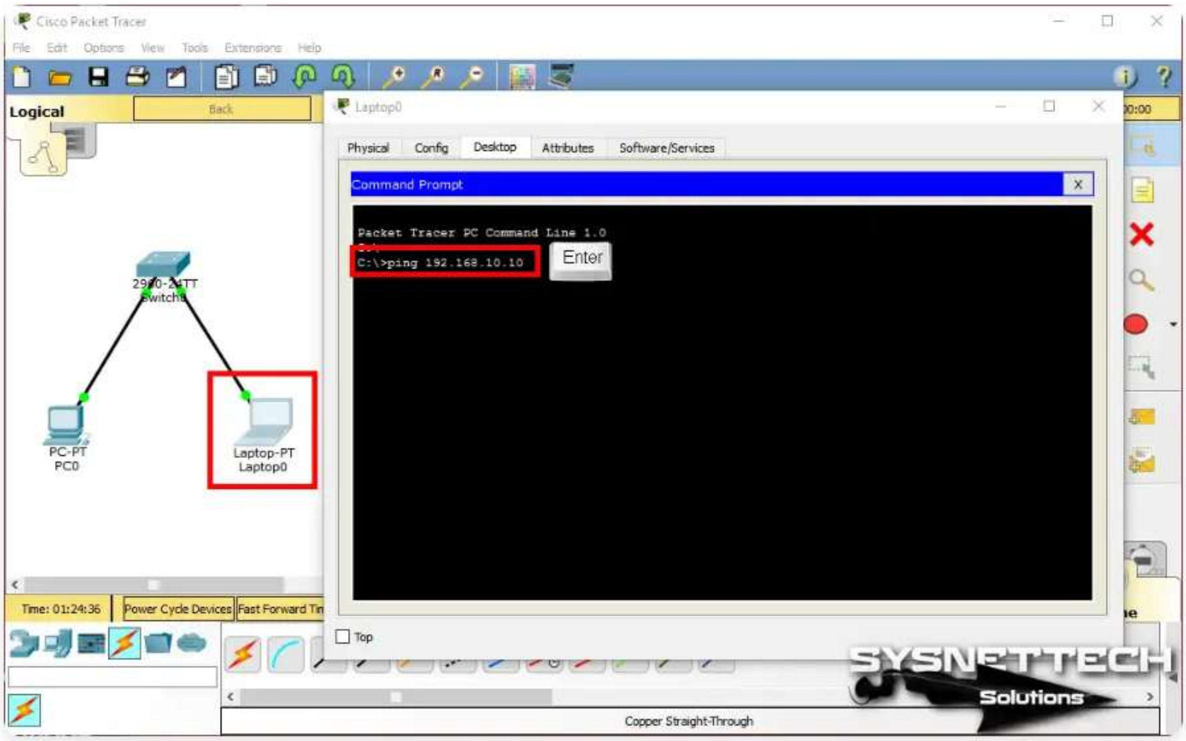


Configure the IP settings of Laptop0 as Static as follows.

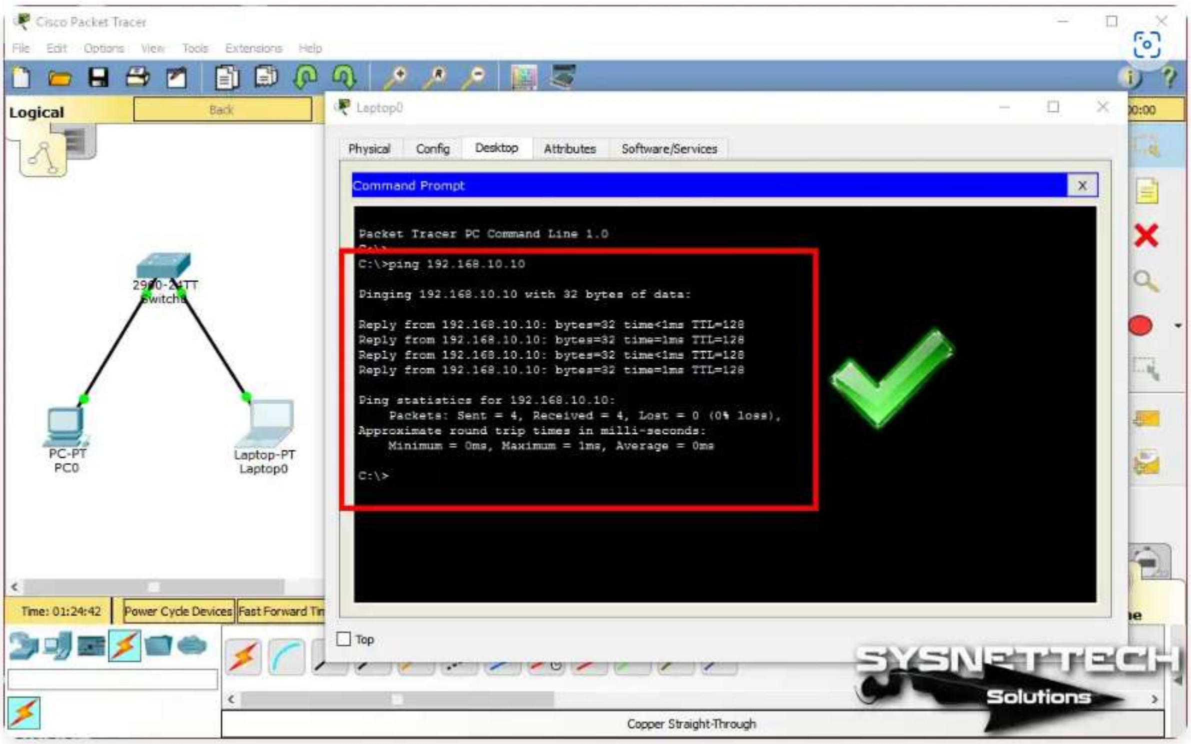


Step 5: How to Test Ping Between Computers

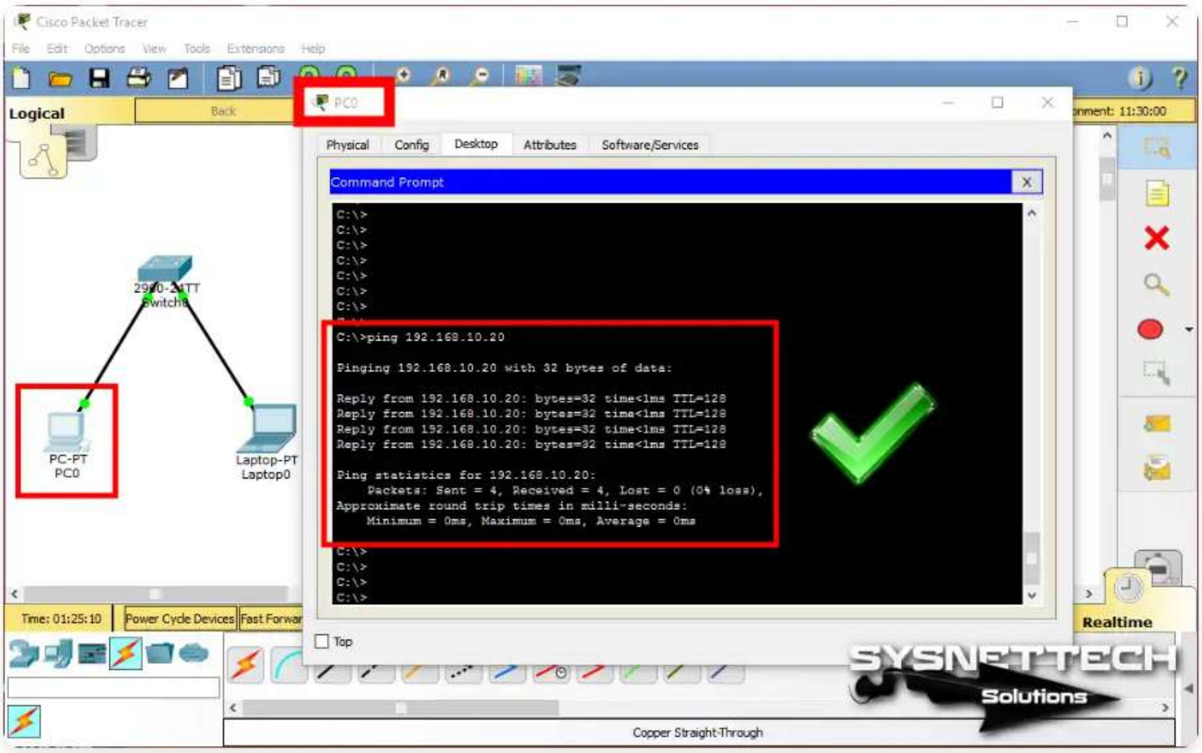
To use ping in PT, you must open the properties of the client device and run the command prompt. Now, to ping from Laptop0 to PC0, execute the ping 192.168.10.10 command on the CMD.



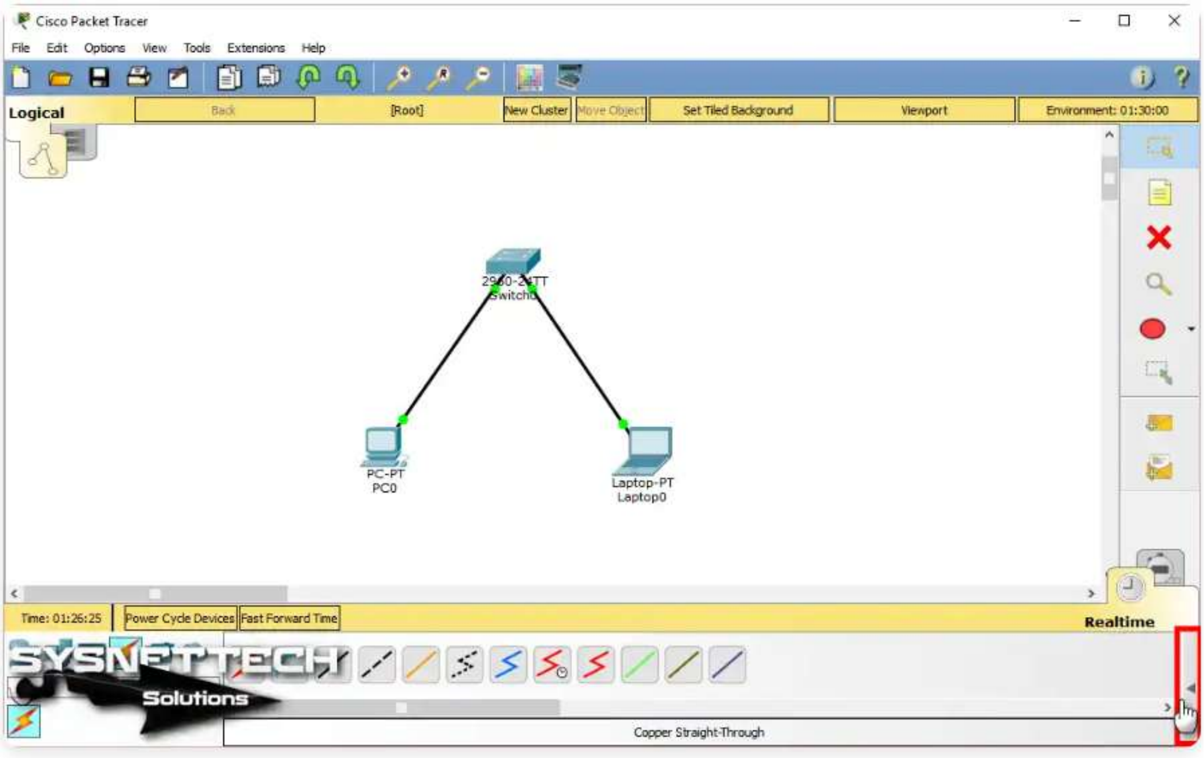
Pinging from Laptop0 to PC0 will be successful as follows.



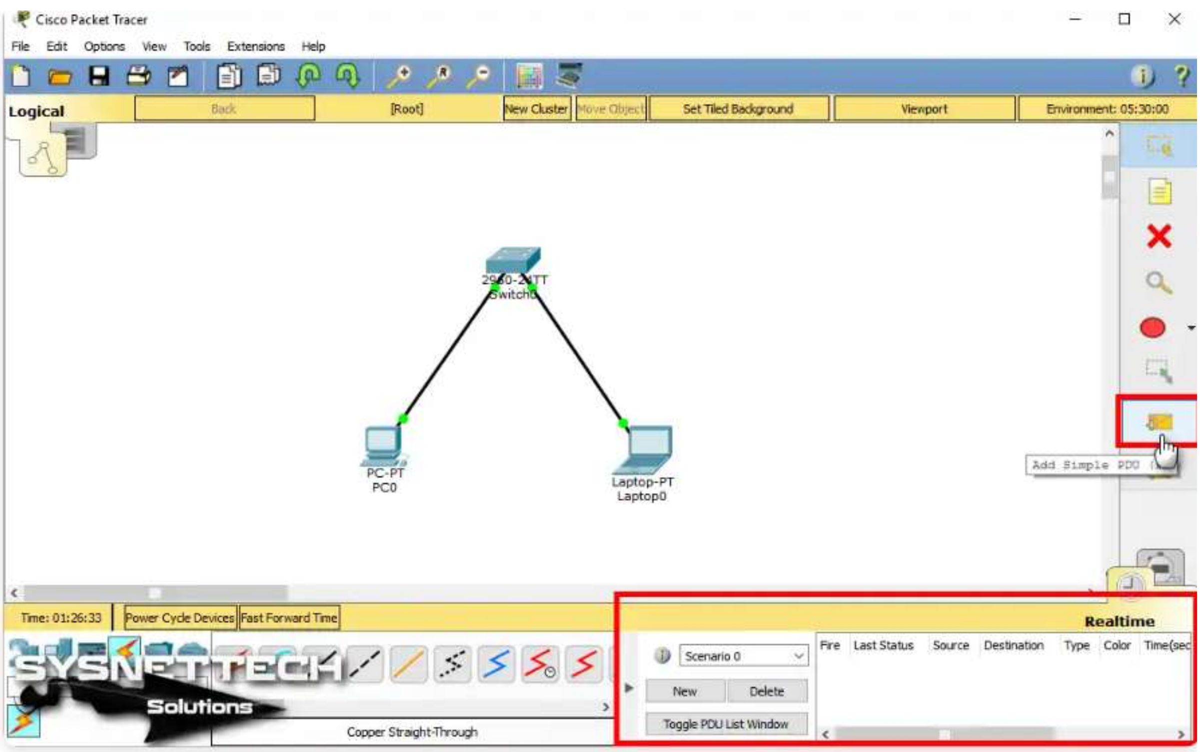
When you ping from PC0 to Laptop0, ping will also be successful.



You can also use Realtime to Ping in the PT. Open Realtime by clicking the left arrow, as you can see below.

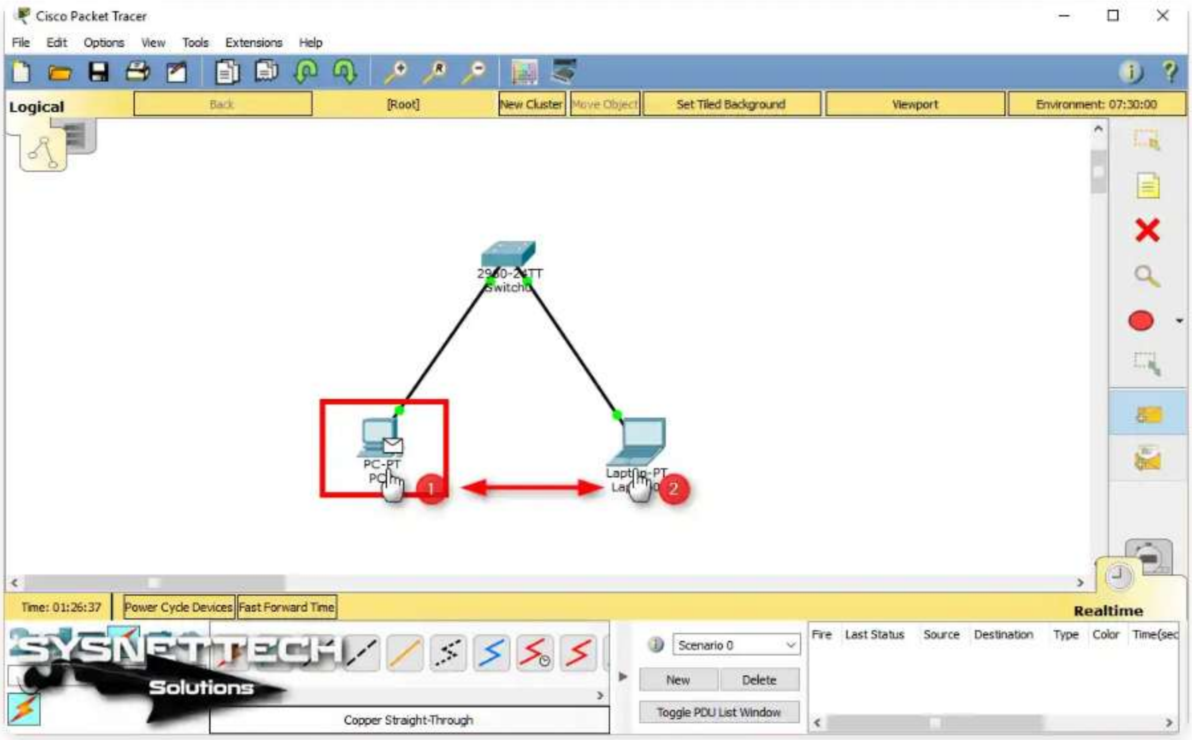


Then click Add Simple PDU in the right pane.

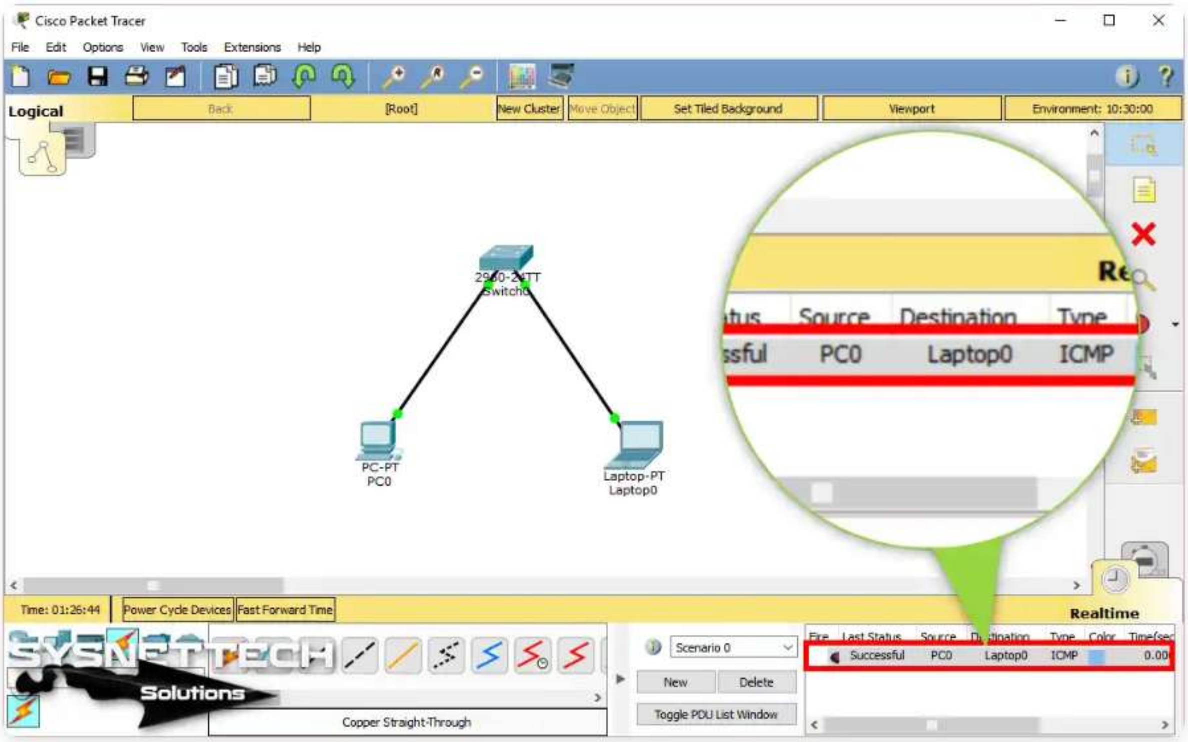


After clicking ADD Simple PDU, you will see an envelope shape. Using this shape, you must select the source and destination client that you want to ping.

Click on PC0 once and then click on Laptop0 to start the Ping process by identifying the source and destination devices.



As soon as you leave the envelope on Laptop0, you will see a message on the Realtime tab that Ping is successful.

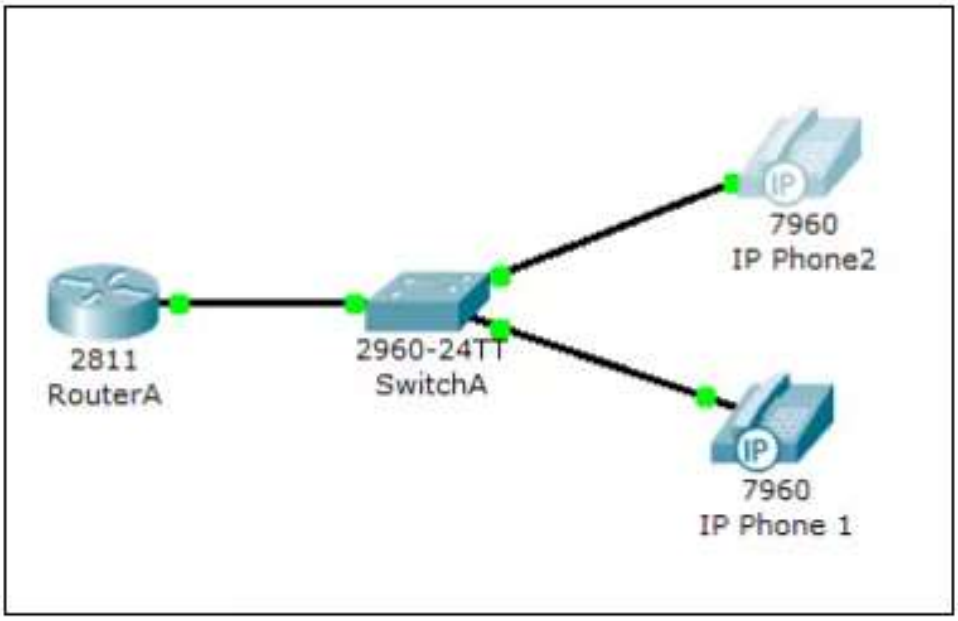


- **eDraw Max**

1. Install eDraw Max on your computer.
2. Launch the software and select the type of diagram you want to create.
3. Use the extensive library of shapes and symbols to build your diagram.
4. Customize the diagram with labels, text, colors, and formatting.
5. Save and export your diagram in various file formats or share it with others.

- **GNS3**

1. Install GNS3 on your computer and set up any additional software or hardware requirements (like virtual machines for network OS).
2. Import or create network topologies by adding virtual network devices, such as routers, switches, and firewalls.
3. Configure the network devices with the desired settings and operating systems.
4. Interconnect devices and create network scenarios.
5. Launch the emulation to test configurations, troubleshoot network issues, and gain hands-on experience with complex network setups.



Points to Remember

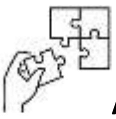
- ✓ **A physical network diagram** illustrates the interconnection of the devices in the network with wires and cables while a logical diagram Illustrates the way information flows through a network. To be precise, the physical network diagram reveals the network topology with all the physical aspects, such as ports, cables, racks, servers, specific models, etc.

- ✓ **Physical diagram in VoIP** system illustrates the interconnection of the devices in the network with wires and cables while a logical diagram Illustrates the way information flows through a network. Where has difference types that are Star topology, Mesh topology and tree (extended) lastly it requires difference tools for making physical diagram. Those tools are: GNS3, eDraw max and packet tracer.

- ✓ **A logical network diagram illustrates** the flow of information through a network and shows how devices communicate with each other. It typically includes elements like subnets, network objects and devices, routing protocols and domains, voice gateways, traffic flow and network segments. In logical network diagrams, there are pivots for small, medium and large networks, where network diagram templates can be helpful.

- ✓ **VoIP network design tools include** Packet tracer, eDraw max and GNS3.

- ✓ **A logical network diagram** illustrates the flow of information through a network and shows how devices communicate with each other. It typically includes elements like subnets, network objects and devices, routing protocols and domains, voice gateways, traffic flow and network segments. For making logical diagram we use packet tracer, eDraw max and GNS3 tools.



Application of learning 1.3. Description of Logical diagram and Use of design tools

Mr Gakire needs to build VoIP system, so there is prerequisite activity required before that activity is to make network diagram for facilitating who will deploy VoIP equipment. As trainee studied VoIP system diagram design, use one design tool, make logical diagram of this system Mr. Gakire needs.

Indicative content 1.4: Estimation of VoIP System Cost

Duration: 2 hrs

Theoretical Activity 1.4.1: Identification of item specifications and items cost

. Tasks:

Task 1: Answer the following questions related to the item specification and item cost

i. What is cost of estimation?
ii. Mention the element/components of cost estimation?
iii. What are item specification and item cost in VoIP system?

Task 2: Provide the answer for the asked questions and write them on papers and present the findings/answers to the whole class.

Task 3: In addition, ask questions where necessary

Task 4: For more clarification, read the key readings 1.4.1.

Key readings: Identification of item specifications and items cost

- **Identify item specifications.**

✓ **Define cost of estimation**

Cost estimation refers to the process of approximating the financial expenses or expenditures required to complete a project, produce a product, or provide a service. It involves assessing and quantifying various cost elements, such as labour, materials, equipment, overhead, and any other resources necessary to accomplish the specified objectives.

Cost estimation is crucial in project management, budgeting, financial planning, and decision-making, as it enables organizations and individuals to allocate resources effectively, set pricing strategies, and determine the feasibility of proposed endeavors. The accuracy and reliability of cost estimates play a significant role in successful project execution and achieving financial objectives.

✓ **Mention the element/components of cost estimation.**

- **Labor Costs**: This encompasses wages and salaries for personnel involved in the project, including project managers, engineers, technicians, and laborers. It may also include employee benefits and payroll taxes.
- **Materials and Supplies**: These are the costs associated with acquiring raw materials, consumables, equipment, and supplies required for the project. This

category also includes the cost of purchasing or renting equipment or machinery.

- **Overhead Costs**: Overhead costs cover indirect expenses necessary to support the project, such as utilities, office space, insurance, and administrative expenses.
- **Subcontractor Costs**: If external contractors or subcontractors are involved, their fees and expenses are factored into the cost estimation.
- **Travel and Transportation Costs**: Expenses for travel, transportation, and logistics, including fuel, vehicle maintenance, and airfare, if necessary.
- **Consulting Fees**: If external consultants or experts are required for specific aspects of the project, their fees and expenses are included.
- **Permits and Licensing Fees**: Costs associated with obtaining permits, licenses, or compliance with regulatory requirements.
- **Software and Technology Expenses**: Costs for software, tools, or technology infrastructure required for the project, including licenses and maintenance.
- **Marketing and Sales Costs**: For product-based projects, these costs include advertising, promotion, and sales-related expenses.
- **Contingency and Risk Management**: A portion of the budget is allocated for unforeseen events or risks that may impact the project's cost. This serves as a buffer to accommodate unexpected expenses.
- **Taxes and Duties**: Any applicable taxes, customs duties, or tariffs are considered when estimating costs.
- **Financing and Interest Costs**: If the project requires borrowing funds or financing, interest expenses may be included in the cost estimation.
- **Inflation and Escalation**: Projections for inflation and cost escalation over the project's duration are factored in.
- **Depreciation**: For long-term projects, depreciation of assets and equipment may be included.
- **Indirect Costs**: These are additional, general costs not directly tied to a specific project, such as corporate overhead.
- **Quality Control and Testing**: Expenses related to quality control and testing to ensure the project's quality and compliance with standards.
- **Environmental and Sustainability Costs**: Costs related to environmental impact assessments, compliance with environmental regulations, and sustainability initiatives.
- **Legal and Compliance Costs**: Expenses for legal services, regulatory compliance, and contracts.
- **Training and Development Costs**: If staff training and development are required for the project, these costs are considered.
- **Management and Contingency Reserves**: These are allowances for project management and unforeseen contingencies.

- **Item Specification**
  - ✓ **Definition**: Item specification refers to a detailed description of a specific component, device, or piece of equipment that is part of the VoIP system. It provides technical details, features, and requirements for the item, ensuring that it meets the needs of the VoIP system.
  - ✓ **Examples**: Item specifications in a VoIP system might include details about IP phones (e.g., model, brand, supported protocols), network switches (e.g., number of ports, power over Ethernet support), servers (e.g., specifications, storage capacity), and software licenses (e.g., number of users, feature set).
    - **Item Cost**
  - ✓ **Definition**: Item cost refers to the monetary value or price associated with the acquisition, purchase, or licensing of a specific item or component for the VoIP system. It represents the financial aspect of acquiring and integrating the item into the system.

Examples: Item costs in a VoIP system might include the purchase price of IP phones, the licensing fees for VoIP software, the cost of network switches, and any additional expenses such as shipping or installation fees.

**Practical Activity 1.4.2: Estimating VoIP system cost**

**Tasks:**

Task 1: Referring to key reading (1.4.1) you are requested to perform the given task. The task should be done individually.
As network technician make cost of estimation of any VoIP and show item specification, item cost and total cost of item by referring to the previous theoretical activity no: 1.4.1.
Task 2: List out procedures and formulas to be used to perform the given tasks.
Task 3: Follow the given instructions.
Task 4: Referring to procedures and formulas provided on task, Perform the given tasks.
Task 5: Present your work to the trainer and whole class.
Task 6: For mire clarification read key readings 1.4.2.

Key readings 1.4.2: Estimating VoIP system cost.

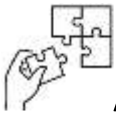
- **Calculate the total cost.**

✓ **Cost of estimation of VoIP System of X organization.**

No	Item	Item specification	Quantity	Cost of item	Total cost
1	Router	SISCO Router x890	1	1,000,000Ffrws	1,000,000Ffrws
2	Switch	SISCO Catalyst 29600 48 ports	2	1,500,000frwa	3,000,000frwa
3	Ip phone	Cisco Ip phone 7821	6	200,000frws	600,000frws
4	IPBX	Max GSM/3G/4G Ports 4	1	450,000Frws	450,000Frws
5	Network cable	Ethernet cable (Cat7)	2 box	150,000frws	300,000frws
6	Cable connector	RAJ11 and RJ 45	100 for RJ45 and 100 for RJ11	100frws	20,000frws
7	Man power	Experience of 3 years	3	50,000frws	150,000frws
8	Transport	Total	1	200,000frws	200,000frws
9	Technician	Experience of 2 years	2	150,000frws	300,000frws
10	Supervisor	Experience of 5 years	1	250,000frws	250,000frws
General Total					**6,270,000frws**

Points to Remember

- ✓ Cost estimation refers to the process of approximating the financial expenses or expenditures required to complete a project, produce a product, or provide a service.
- ✓ Element/components of cost estimate include Labor Costs, Materials and Supplies, Overhead Costs, Subcontractor Costs, Travel and Transportation Costs, Consulting Fees, Permits and Licensing Fees, Software and Technology Expenses.
- ✓ Estimation of VoIP system Cost comprise of identification of item specification (Item specifications in a VoIP system might include details about IP phones), cost of item (Item cost refers to the monetary value or price associated with the acquisition, purchase, or licensing of a specific item or component for the VoIP system. It represents the financial aspect of acquiring and integrating the item into the system.) and total cost (the total budget that will spend to the items).



Application of learning 1.4:

XXXX Telecom Company needs to install VoIP system in the new office located at Musanze city, this company needs to know the required tools, equipment and materials and their cost, so as VOIP technician you are hired to prepare cost estimation of this project.

Do this:

a) Identify all item required.
b) Identify the item specification.
c) Identify the cost of each item.
d) Calculate the cost total cost.
e) Print the cost estimation.

Learning outcome 1 end assessment

Theoretical assessment

Q1. Choose the correct answer.

i. What is the primary goal of network assessment in VoIP system planning?

A. Identify potential security threats.
B. Determine the optimal number of VoIP phones.
C. Evaluate network infrastructure for VoIP suitability.
D. Calculate the total cost of ownership (TCO)

ii. Which of the following factors is NOT typically considered when determining bandwidth requirements for a VoIP system?

A. Number of simultaneous calls
B. Codec used for audio compression.
C. Network topology
D. Office location

iii. What is the main purpose of a Quality of Service (QoS) implementation in a VoIP network?

A. Improve call quality.
B. Enhance network security.
C. Reduce network latency.
D. Increase network bandwidth.

Q2. Answer by true or false

i) A VoIP system can operate independently of a data network.
ii) Power over Ethernet (PoE) can be used to supply power to VoIP phones.
iii) Call detail records (CDRs) provide valuable information for network troubleshooting.

Q3. Match the following VoIP system components with their descriptions:

SN	System components	Descriptions	
1	VoIP Gateway	i)	Converts analogy signals to digital and vice versa
2	Media Server	ii)	Handles internal and external phone calls within an organization
3	SIP Server	iii)	Connects analog phones to a VoIP network

4	Codec	iv)	Manages call setup and termination
5	ATA (Analog Telephone Adapter)	v)	Compresses and decompresses audio data
6	PBX (Private Branch Exchange)	vi)	Stores voice messages

Practical assessment

WXY Business group have project of deploying new VoIP system to the headquarter office located at KAGARAMA and they need to hire expert technician who will be responsible to make field survey, making VoIP system network diagram (Physical and logical diagram), and to estimate cost of all required equipment and materials.

As technician you will hire to do those tasks.

a. Select tools, equipment and materials you need for this project.
b. Conduct field survey (physical survey) of this project.
c. Provide blueprint of this project
d. Design the VoIP system diagram (physical and logical diagram) in this project.
e. Estimate the cost of this project.

This assessment will take 2hours.

END

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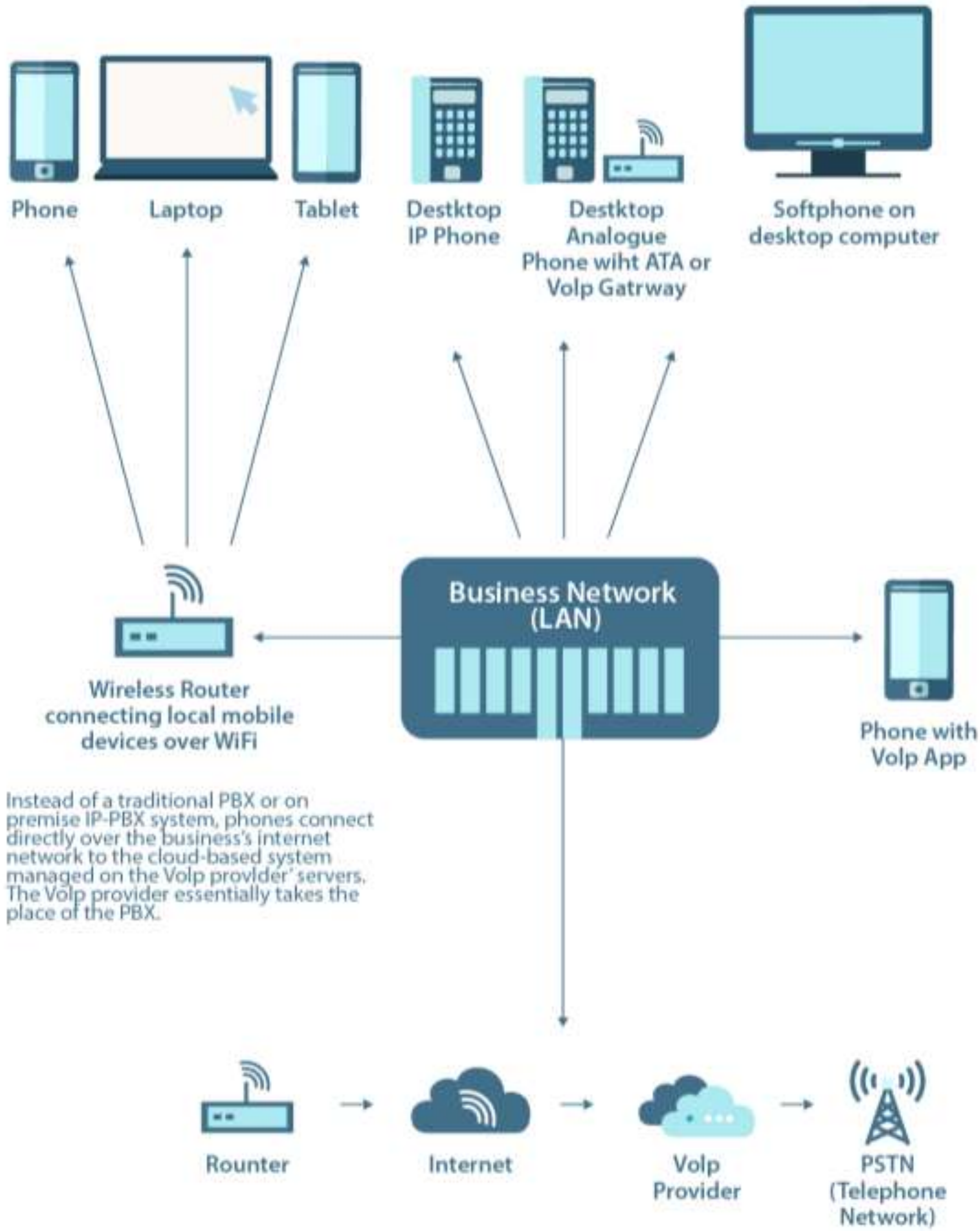

Phone
Laptop
Tablet
Destktop IP Phone
Destktop Analogue Phone wiht ATA or Volp Gatrway
Softphone on desktop computer
Business Network (LAN)
Wireless Router connecting local mobile devices over WiFi
Phone with Volp App
Instead of a traditional PBX or on premise IP-PBX system, phones connect directly over the business's internet network to the cloud-based system managed on the Volp provider' servers. The Volp provider essentially takes the place of the PBX.
Rounter
Internet
Volp Provider
PSTN (Telephone Network)

Indicative contents **2.1. Selection of VoIP system Tools, materials and equipment** **2.2. Perform VoIP system trunking.** **2.3. Connection of VoIP devices**

Key Competencies for Learning Outcome 2: Deploy VOIP system equipment.

Knowledge	**Skills**	**Attitudes**
• Description of tools used in VoIP system. • Explanation of materials for VoIP system and Differentiation of material to another. • Description of equipment's used in VoIP system. • Identification of VoIP system trunking • Illustration on connection of VoIP system	• Using VoIP tools • Selecting Material used in communication. • Using the equipment of VoIP system • Protecting VoIP cables • Connecting VoIP system	• Being attentive • Having teamwork • Having Self-motivation • Being Confident

Duration: 10 hrs

Learning outcome 2 objectives:

By the end of the learning outcome, the trainees will be able to:

1. Describe properly tools based on VoIP system requirements.
2. Use correctly tools based on VoIP system requirements.
3. Describe properly materials based on VoIP system requirements.
4. Select correctly materials based on VoIP system requirements.
5. Describe properly equipment based on VoIP system requirements.
6. Use correctly equipment based on VoIP system requirements.
7. Identify appropriately Piping (Trucking) according to the VoIP system design.
8. Protect properly VoIP cables according to VoIP system design.
9. Illustrate properly VoIP system based on VoIP design and user manual guideline.
10. Connect properly VoIP devices based on VoIP design and user Manual guideline.

Resources

Equipment	**Tools**	**Materials**
• Router • PBX • APBX • Phone • Tapes measures • Switch • Computer	• Designing tools (Packet Tracer, eDraw max, GNS3) • Networking tool kit • Calculator	• Internet bundles • Network cable • Cable connector

Indicative content 2.1: Selection of VoIP System Tools, Materials and Equipment

Duration: 4 hrs

Theoretical Activity 2.1.1: Description of Tools, Materials and Equipment's of VOIP System

Tasks:

Task 1: Answer the following questions related to the description of tools, materials and equipment's of VOIP System.

i. Which tool is used for cutting, drilling, patching and testing?
ii. How those tools can be used?
iii. What is the use of every material needed for VoIP system?
iv. What is the function of each equipment used in VoIP system?

Task 2: Provide the answer for the asked questions and write them on papers.
Task 3: Present your findings to the whole class.
Task 4: In addition, ask questions where necessary.
Task 5: For more clarification, read the key **readings 2.1.1**

Key readings 2.1.1.: Description of Tools, Materials and Equipment's of VOIP System

- **Tools**

A tool is something you use to do something, while **materials** are consumable items on which you use tools to make, achieve and **Equipment** refers to a set of tools or other objects commonly used to achieve a particular objective.

✓ **Cutting tools**

A cutting tool is any tool that is used to remove material from the workpiece by means of shear deformation. Cutting may be accomplished by single-point or multipoint tools.

Wire Cutter: - To cut the network cable of the required length from the bundle, you can use any standard wire cutter tool or can use a wire cutter tool that is specially designed for the twisted-pair cable. A twisted-pair wire cutter usually includes additional blades for stripping the wire.

Wire Stripper: - This tool is used to remove the outer and inner jackets of the network cable. Typically, you do not need to purchase this tool separately as all standard twisted-pair wire cutters are equipped with wire-strippers. The following image shows two twisted-pair wire cutters.

- ✓ **Drilling tools**

A drill is a tool primarily used for making round holes or driving fasteners. Drills are commonly used in woodworking, metalworking, machine tool fabrication, construction and utility projects.

A hand drill is the simplest form of drills. They are ideal for predrilling holes before putting the screws inside.

- ✓ **Fixing tools**

This metal tool is *specifically designed to tighten cable ties and snip off the excess tail*

CABLE TIE TENSIONING TOOL

- ✓ **Testing tools**

A VoIP test helps you review the overall health of your network connection within a matter of seconds. It also enables you to determine whether the call traffic in your existing phone system is high enough to justify an upgrade.

Top 7 VoIP Testing Tools

- ZDNet Broadband Speed Test.
- Nextiva VoIP Speed Test.
- Freeola Broadband Line Quality Test.
- Speed Matters Speed Test.
- 8×8 VoIP Readiness Test.
- Ping-test.net.
- MegaPath Speed Test Plus.
- OnSIP VoIP Test

- ✓ **Patching tools**

A patch panel in a local area network (LAN) is a mounted hardware assembly that contains ports used to connect and manage incoming and outgoing LAN cables. Patch panels are also referred to as patch bays, patch fields or jack fields and are also commonly used in radio and television.

- **Materials**

✓ **Connectors (Circuit Telephony Connectors)**

The Circuit Telephony Connectors allow you to assign a public or a private directory number to Circuit users, so that they can make and receive phone calls in Circuit, as well as collaborate with non-Circuit users.

There are five types of Telephony Connectors:

1. Hosted Universal Telephony Connector (hUTC)
2. Premise Universal Telephony Connector (pUTC)
3. Advanced Telephony Connector (ATC)
4. OpenScape Business Connector
5. Subscriber Telephony Connector (STC)

Hosted Universal Telephony Connector (hUTC)

The hUTC is an application running in the cloud that allows you to connect your SIP-based platform to Circuit using a generic SIP trunk. It requires the following:

- A static SIP trunk must be established between your voice platform and Circuit.
- Call routing on your voice platform must be configured.

The hUTC has a limited size (i.e., less than 100 users per Circuit domain) and therefore it is best suited for small and medium-sized business or for initial customer field trials

Premise Universal Telephony Connector (pUTC)

The pUTC allows you to provide secure communication between the OpenScape Session Border Controller (OpenScape SBC (A *session border controller* is like a firewall specifically designed for VoIP)) and Circuit, using a secure Web Socket for signaling and UDP/ DTLS (Datagram Transport

Advanced Telephony Connector (ATC)

The Advanced Telephony Connector (ATC) provides a richer set of features in comparison with the Universal Telephony Connectors. This functionality requires an additional interface to the PBX which is only supported on the Unify PBXs, OpenScape Voice and OpenScape

4000.The ATC allows to:

1. use One Number Service (ONS), i.e. one phone number and:
   - receive calls at the office phone number, so that no new phone number is required.
   - accept an incoming call on all devices, such as web or desktop app, mobile app or desk phone.
   - send ONS / office phone number as calling party.
2. Be mobile and move your calls to your needs:
   - use Circuit clients as softphones.
   - pull call from other Circuit clients or desk phone.
   - push call to desk phone or alternative phone, e.g. mobile phone
   - fallback option for softphone users that are not logged in (allows to hand over to a predefined alternative number, e.g. mobile phone)

- **OpenScape Business Connector**

The OpenScape Business Connector allows you to easily connect your OpenScape Business PBX V2 (with a Service Releases 2 or higher) to Circuit.

A Circuit domain-specific API stands for Application Programming Interface (is tool for software developers to make and receive phone calls with a simple, easy to understand) key is entered into your opens cape Business system. This is needed to establish a SIP trunk between Opens cape Business and Circuit and allow maintenance from the WBM (Web Based Administration).

Call routing is configured on Opens cape Business and Opens cape Business number is linked to the Circuit user.

The Opens cape Business Connector (with Opens cape Business V2R6 FR2 and higher) allows to:

use One Number Service (ONS), i.e. one phone number and:

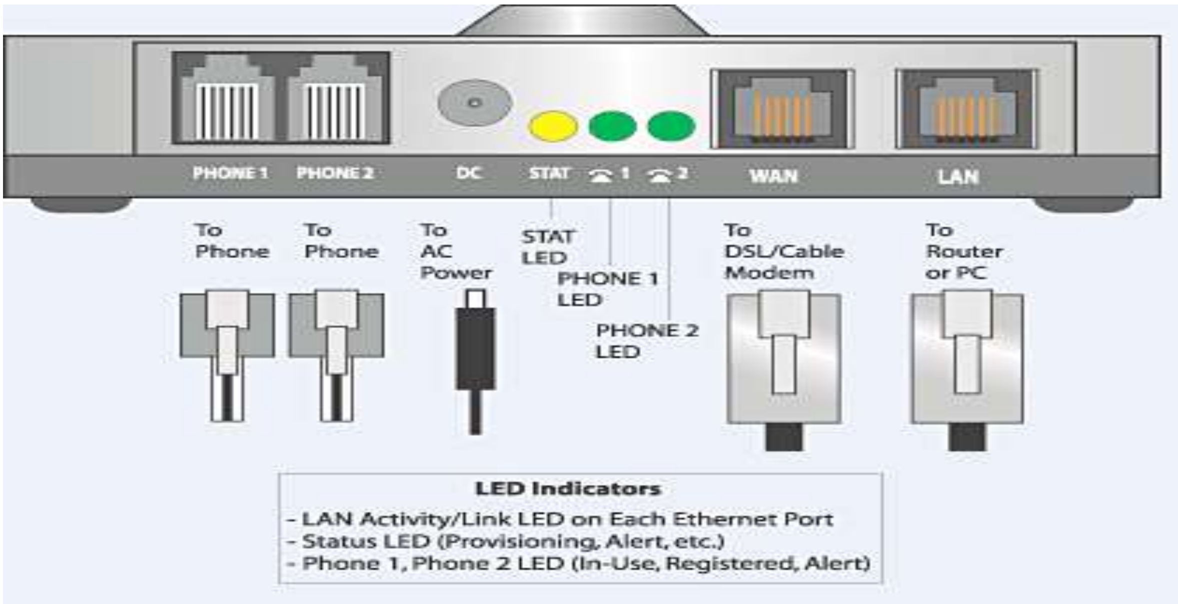
- receive calls at the office phone number, so that no new phone number is required.
- accept an incoming call on all devices, such as web or desktop app, mobile app or desk phone.

- **Subscriber Telephony Connector (STC)**

The Subscriber Telephony Connector (STC) provides a richer set of features and easier setup in comparison with the Universal Telephony Connectors. Note: In most of the cases, the deployment of a Circuit Telephony Connector requires special configuration on the firewall.

✓ **Analog telephone adapter**

An analogy telephone adaptor (ATA) is a device used to connect a standard telephone to a computer or network so that the user can make calls over the Internet. Internet-based long distance calls can be substantially cheaper than calls transmitted over the traditional telephone system, and ATAs are typically cheaper than specialized VoIP phones that connect directly to a computer's Universal Serial Bus (USB) port.



✓ **Cable**

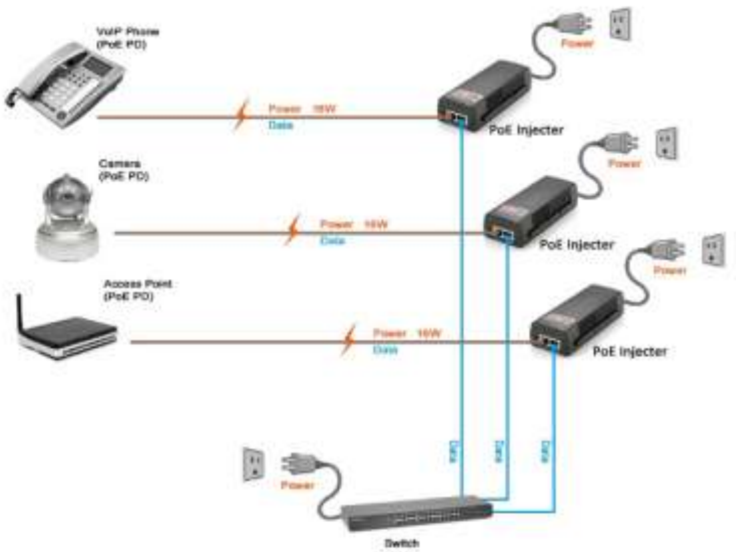
There are two types of cables associated with Voice Over IP connectivity:

Telephone wire and Ethernet cables. Telephone cable comes in two common varieties, the "line cord" and the "handset cord." Ethernet cables are commonly called "Cat5(**a deliver Gigabit Ethernet speeds of up to 1000Mbps**)" or "Cat6" cables, which causes some confusion. Cat6 cables are newer and better than Cat5, but, if the cables weren't labelled, they'd look the same. What cable is used for VoIP?

Each workstation where you will be installing a VoIP phone must have **Category 5 (CAT5), Category 5 Enhanced (CAT5E), or Category 6 (CAT6)** cabling installed with an Ethernet (RJ45) jack. CAT6 cabling should be used where throughput of greater than 100Mbps is desired.

- ✓ **PoE adapter**

Power over Ethernet (PoE) is the technology that integrates data, voice and power over standard Ethernet infrastructure using Cat 5 or better cables. It is the means to supply reliable, uninterrupted power to Internet Protocol (IP) network cameras, IP phones, WLAN Access points, Thin Clients and other Ethernet devices.



- ✓ **Jumper cable**

Jumper Cable means a pair of twisted insulated conductors that connects from NBN (national broadband network) side of the main distribution frame to your side of the main distribution frame in an MDU (the gateway converts each local analogy phone circuit into a SIP).

- **Equipment**

✓ **Channel bank**

A channel bank is a device at a telephone company central office (public exchange) that converts analogy signals from home and business users. otherwise, A device that performs multiplexing or demultiplexing of a group of communications channels into one channel of higher bandwidth or higher digital bit rate. The D-type Channel Bank is used for digital signals. There are five kinds of Channel Banks that are used in the System: D1, D2, D3, D4, and DCT (Digital Carrier Trunk).

What is a T1 channel bank?

In telecommunications, a channel bank is a device that performs multiplexing or demultiplexing ("demux") of a group of communications channels, such as analog or digital telephone lines, into one channel of higher bandwidth or higher digital bit rate, such as a DS-1 (T1) circuit, so that all the channels can be sent ...

✓ **Ip phone**

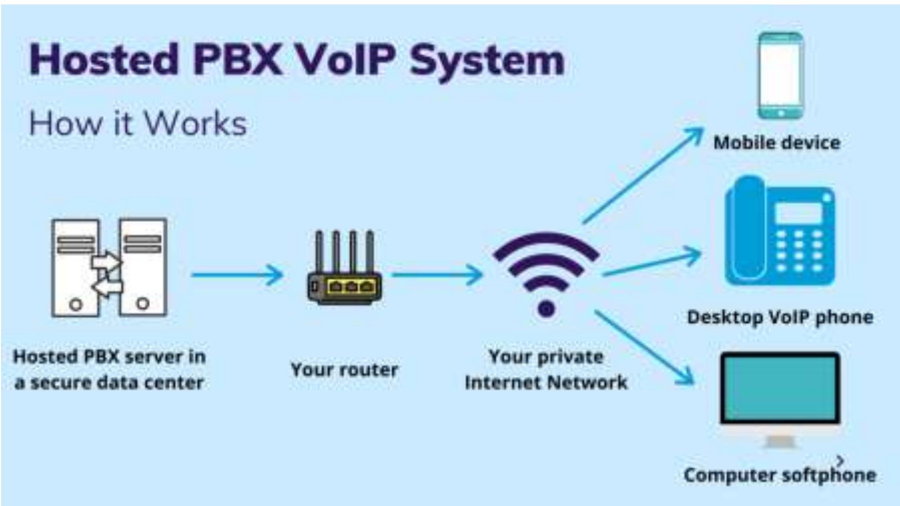
A VoIP phone is a hardware- or software-based telephone designed to use voice over Internet Protocol (VoIP) technology to send and receive phone calls over an IP network. The phone converts analogy telephony audio into a digital format that can be transmitted over the internet and converts incoming digital phone signals from the internet to standard telephone audio.

- ✓ **Power backup (UPS)**

Backup power is defined as any device that provides instantaneous, uninterruptible power, VoIP phone systems need electricity and internet to function. In the event of a power outage, a battery backup unit will give you the opportunity to keep making calls for a short period of time. In the event of a total blackout, internet may not be available because the Internet service provider will also lose power.

- ✓ **IP PBX Server/PBX Server**

An IP PBX is a private branch exchange (telephone switching system within an enterprise) that switches call between VoIP (voice over Internet Protocol or IP) users on local lines while allowing all users to share a certain number of external phone lines.



- ✓ **VoIP Router**

is a specialized router that allows you to make phone calls over the Internet By using a router for VoIP, you can receive incoming calls through your Internet connection. VoIP routers are what let you make those phone calls over an internet connection from your internet service provider, rather than traditional telephony methods. There are both dual-band (two signals) and tri-band (three signals) routers, depending on what speeds you need.

- ✓ **VoIP Gateway**

A VoIP gateway is a gateway device that uses Internet Protocols to transmit and receive voice communications.

- ✓ **VoIP Headset**

Connect over a telephone or to a computer, allowing the user to speak and listen while keeping both hands free.

- ✓ **PoE Switch**

Power over Ethernet (PoE) is the technology that integrates data, voice and power over standard Ethernet infrastructure using Cat 5 or better cables. otherwise, Power over Ethernet (PoE) is a technology that allows network switches to transmit both power and data through an Ethernet cable simultaneously. The main difference between PoE and non-PoE switches is that a PoE switch uses Power over Ethernet technology to deliver data and power to devices, whereas a non-POE switch only delivers data.

Points to Remember

- ✓ Selections of tools depends on which tool you are going to use VoIP work; We have cutting tools, drilling tools, fixing tools, patching tools and testing tools.
- ✓ By selecting the VoIP material consider the following Connectors, analogy telephone adapter, Cable, PoE adapter, Patch code and Jumper cable.
- ✓ The VoIP equipment's are channel bank, Ip phone Power, backup (UPS), IP PBX Server/PBX Server, VoIP Router, VoIP Gateway, VoIP Headset PoE Switch.

Application of learning 2.1.

As a technician in network engineering industry there is a client who has problem while making voice calls, when He is on phone call, He is not able to get the voice from the receiver on the call, now you are requested to select the needed materials, tools and equipment.

Indicative content 2.2: Perform VoIP System Trunking.

Duration: 3 hrs

Theoretical Activity 2.2.1: Description of installation Types

Tasks:

1: Answer the following questions related to the Description of VoIP installation Types.

i. Explain system trunking and installation types.
ii. What is difference between plastic and stainless steel?
iii. What are VOIP equipment's that needs assemble?

2: Provide the answer for the asked questions and write them on papers.

3: Present the findings/answers to trainer and the whole class.

4: In addition, ask questions where necessary.

5: For more clarification, read the key readings 2.2.1

Key readings 2.1.1.: Description of installation Types

SIP Trunking refers to the backbone of phone lines used by multiple users that connects to a telephone network.

- **Description of installation Types**

✓ Visible
✓ Built in
✓ Semi built in
✓ Waterproof
✓ Underground

- **Description Type of trunking**

✓ **Plastic** refers to the protection of cables from damage and helps to keep your wires tidy.

- ✓ **Stainless steel** is an alloy of iron that is resistant to rusting and corrosion.

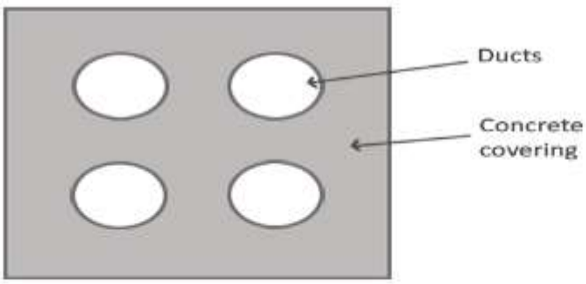
- **Application of Cable Laying and pulling methods**
  - ✓ **General methods**

Cable Laying The reliability of underground cable network highly depends upon proper laying of cables, quality of cable joints and branch connections etc. There are three main **methods of laying underground cables**, which are - (i) direct laying, (ii) draw-in system and (iii) solid system.

- ✓ **Cable laying arrangement**

Direct laying of underground cables the direct system of cable laying involves the cables laid in a cable trench and after that, the cable is filled with soil.

Draw in system cable layout methods is widely used for heavily populated areas such as urban cities and towns. Normally the cables are layout in ducts or pipelines on the ground with certain holes.

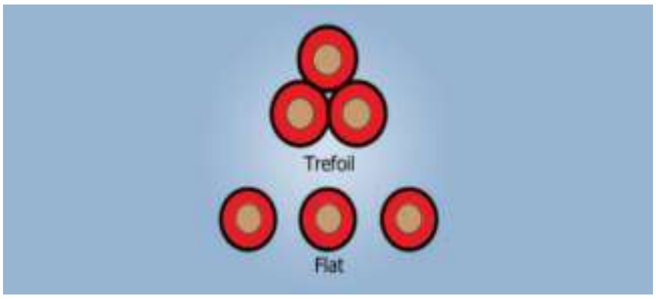


Solid System The solid system of cable laying is mainly performed where the cable is laid in the wood, cast iron, or another non soil surface.

✚

Cable laying arrangement

Multicore cables are laid in "flat formation" arrangement and single core cables may be laid in "trefoil" arrangement or in "flat formation" arrangement.



- ✓ **Segregation of cables Cable**

No matter what method is used for laying, cables must be segregate taking into account the voltage level and function, to avoid possible electromagnetic! interferences that can disturb the networks and the signals cables are carrying on.

- ✓ **Cable Marking**

Cable marking is the process of identifying wires, cables, and their respective functions within a system. It's a crucial aspect of maintaining and troubleshooting electrical and electronic equipment. Effective cable marking ensures clarity, safety, and efficiency.

Common methods of cable marking include:

- **Colour coding:** Using different colours to distinguish wires.
- **Heat shrinks tubing:** Applying printed heat shrink sleeves onto cables.
- **Cable labels:** Attaching adhesive labels with printed information.
- **Wire markers:** Small plastic or metal tags with imprinted text.

- **Assemble VoIP equipment.**

Assembling VoIP equipment typically involves connecting hardware components and configuring software settings to establish a functional VoIP system. While the complexity can vary depending on the system's size and features, here's a general outline:

- ✓ **Essential Components:**

- **VoIP phones:** These are specialized phones designed for VoIP networks.
- **Router:** This device connects your network to the internet.
- **Ethernet cables:** These connect the VoIP phones to the router or a switch.
- **VoIP service provider account:** This provides the necessary credentials for making and receiving calls.

✓ **Assembly Steps:**

1. **Connect hardware:**
    - Connect the VoIP phones to the router using Ethernet cables. Ensure you're using the correct ports.
    - Connect switch to the router and then connect the VoIP phones to the switch.
2. **Configure VoIP phones:**
    - Refer to the phone's user manual for specific instructions.
    - Enter the VoIP service provider's information, including username, password, and server address.
3. **Configure router:**
    - Configure the router to prioritize VoIP traffic for optimal call quality. This often involves Quality of Service (QoS) settings. Check your router's manual for specific instructions.
4. **Activate VoIP service:**
    - Contact the VoIP service provider to activate the service and obtain necessary credentials.
5. **Test the system:**
    - Make test calls to ensure everything is working correctly. Check call quality, audio clarity, and feature functionality.

✓ **Rack mount**

A rack mount for a VoIP system is essential for organizing and managing VoIP equipment effectively, especially in larger installations. It provides a structured environment, improves cable management, and ensures optimal performance.

Key Components for a Rack-Mounted VoIP System

- **VoIP Gateway:** This is the core component that converts analog signals to digital and vice versa. Many VoIP gateways are designed to be rack-mountable.
- **IP PBX:** If your VoIP system uses a private branch exchange, a rack-mountable IP PBX is ideal.
- **Network Switches:** To connect multiple devices, a rack-mountable switch is necessary.
- **Servers:** If your VoIP system relies on servers for additional functionalities like call recording or CRM integration, they can also be rack-mounted.
- **Power Distribution Units (PDUs):** To manage power distribution and monitoring within the rack.

Benefits of Rack Mounting VoIP Equipment:

Space optimization: Efficiently utilizes rack space.

- **Improved airflow:** Proper ventilation prevents overheating.
- **Cable management:** Keeps cables organized and reduces clutter.
- **Security:** Protects equipment from physical damage.
- **Scalability:** Easily add or remove components as needed.

Image of a rack-mounted VoIP system

- ✓ **IP PBX Server/PBX/PABX Server**

An **IP PBX server** is the central component that makes a VoIP system functional. It acts as the brain behind the operation, managing all communication within and outside the organization.

Key Roles of an IP PBX Server in a VoIP System:

- **Call Management:** Handles incoming and outgoing calls, routing them to the appropriate extensions or external lines.
- **Feature Enablement:** Provides a range of features like voicemail, call forwarding, call conferencing, call waiting, and more.
- **Integration:** Connects with other business applications like CRM, email, and unified messaging platforms.
- **Scalability:** Allows for easy expansion as your business grows by adding new users or features.
- **Cost-Efficiency:** Reduces telecommunication costs by utilizing internet connectivity for voice calls.

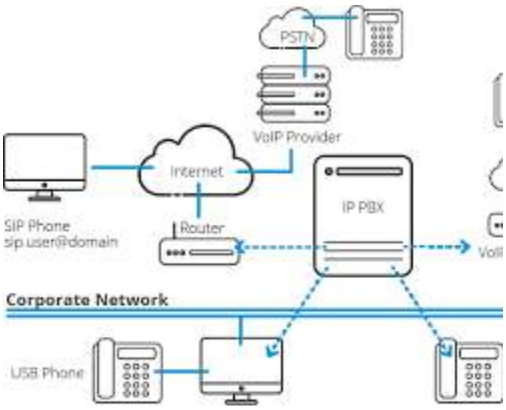
How it Works

- **Call Initiation:** When a user makes a call, the IP PBX server processes the request and establishes a connection.
- **Call Routing:** The server determines the destination of the call, whether it's an internal extension or an external number.
- **Media Handling:** It handles the conversion of voice data into digital packets and vice versa.
- **Feature Activation:** If requested, the server activates features like voicemail, call forwarding, or conference calling.

Types of IP PBX Servers

- **On-premises IP PBX:** Installed within your organization's network, offering greater control but requiring hardware maintenance.
- **Cloud-based IP PBX:** Hosted by a third-party provider, offering flexibility and reduced upfront costs.

A simplified diagram of a VoIP system with an IP PBX server



✓ **VoIP Phone**

A **VoIP phone** is a device specifically designed for making and receiving phone calls over an internet connection. Unlike traditional analogy phones, VoIP phones convert your voice into digital data packets for transmission over a network.

Key Features of VoIP Phones

- **Digital display:** Shows caller ID, call duration, and other information.

- **Ethernet port:** Connects to a network for internet access.
- **HD audio:** Provides clear and high-quality sound.
- **Additional features:** Many VoIP phones offer advanced features like call forwarding, voicemail, and conference calling.

Types of VoIP Phones

- **Desk phones:** Traditional-looking phones with modern technology inside.
- **Softphones:** Software applications that turn computers or smartphones into VoIP phones.

Image of a VoIP phone:

How VoIP Phones Work

- **Voice conversion:** Your voice is converted into digital data.
- **Packet creation:** The digital data is divided into packets.
- **Network transmission:** Packets are sent over the internet to the recipient's VoIP phone.
- **Packet reassembly:** The recipient's phone reassembles the packets and converts the data back into voice.

Points to Remember

- ✓ **Cable Marking**
- ✓ **SIP Trunking** refers to the backbone of phone lines used by multiple users that connects to a telephone network.
- ✓ **Description of installation Types include** Visible, built in, Semi built in, Waterproof and underground.

Application of learning 2.2.

ABCD organization has a growing and it needs to expand its telecommunication capabilities. The current VoIP system is unable to handle the increased call volume and requires additional external lines. You have been tasked with configuring VoIP system trunking to connect the internal PBX to the public switched telephone network (PSTN).

Task 1: Establish a reliable VoIP trunk between the internal PBX and a chosen VoIP service provider.

Task 2: Configure trunk parameters, including dial plans, codecs, and quality of service (QoS).

Task 3: Test the trunk functionality to ensure proper call termination and origination.

Task 4: Troubleshoot any issues that may arise during the configuration process.

Indicative content 2.3: Connection of VoIP Devices

Duration: 3 hrs

Theoretical Activity 2.3.1: Connection of VoIP devices

Task:

Task 1: In small groups, you are requested to answer the following questions related to the Connection of VoIP devices

i. What are the primary requirements to consider before choosing a VOIP device?
ii. What do you understand by network fibre connector?
iii. Why is it necessary to test system connectivity?

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to trainer and the whole class

Task 4: For more clarification, read the key **readings 2.3.1.**

Key readings 2.3.1: Connection of VoIP devices

- **Deployment of VoIP devices**

Primary requirements to consider before choosing a VOIP device.

1. **Look at the brands.** Are there names you trust more than others? Think about longevity and past performance.

Poly is the global leader when it comes to productive and effective engagement with collea gues, partners, customers, and prospects. You can't look at a serious VoIP solution without including its product line in your decision-making process.

Cisco is a big name in business networks and internet infrastructure and another lead playe r in the **VoIP hardware** realm.

Panasonic is one of the world's leading innovators in VoIP phone business systems. Their re putation for making quality electronic devices goes back decades.

VTech is the world's largest manufacturer of **cordless phones** and well known for its electr onic educational toys.

2. **Look for the number of line appearances.** How many lines does each device in your business need to be able to handle? Some team members will need many more than two lines.
3. **Do use headsets?** What type of connector do they have? An RJ-9 is a familiar type o f connector for telephones but is less popular. Today's headsets use USB for connec tivity. If you need Bluetooth, then make sure your device supports it. Some devices

require a USB-to-Bluetooth dongle that might not be included.

4. **Ready for Power over Ethernet (PoE)?** Most devices now support PoE but ensure that you have the power adapters since most employees that work from *home* do not have PoE-capable network equipment.
5. **Don't forget about reliability.** Does the phone receive regular security updates? VoIP phones that receive these updates tend to provide much more **reliable VoIP phone service** than devices that need firmware delivered by hand. (Hint: Many, many internet-connected devices *never* get updated.)

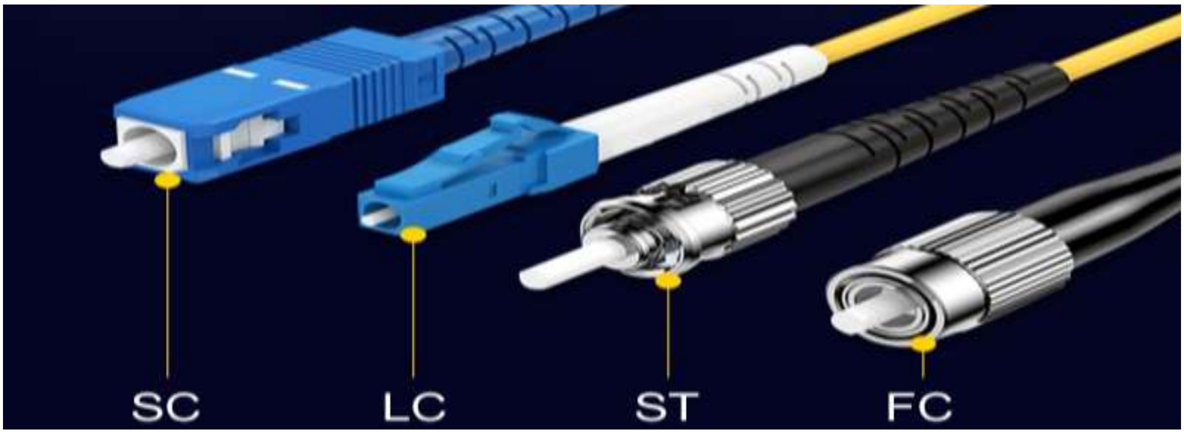
- **Connection of connectors**

✓ **Network Fiber Connectors**

Straight -Tip (ST): An older bayonet style connector widely used with multimode fibre.

Subscriber Connector (SC): Sometimes referred to as square connector or standard connector. It is a widely adopted LAN and WAN connector that uses a push-pull mechanism to ensure positive insertion. This connector type is used with multimode and single-mode fibre.

Lucent Connector (LC): Sometimes called a little or local connector, is quickly growing in popularity due to its smaller size. It is used with single-mode fiber and also supports multimode fibre.



✓ **Coaxial connector**

used to connect cables to other devices and are specifically designed to maintain the shielding on the cable.

- ✓ **Twisted pair connectors.**

The twisted-pair connector interface, often referred to as an RJ-45, is **an 8-position.**

- **Selection of Patching/Tagging components**

- ✓ **Patch panel types**

Twisted-pair copper, These panels are designed for specific twisted-pair copper specifications, like Cat5E, Cat6, Cat6A and Cat7 cables.

Fiber optics. Patch panels are available for both single- and multimode fibre cabling.

Coax patch panels connect devices such as television media players and video cameras to centralized AV switching and mixing systems.

- ✓ **Patch code types**

Patch is a software update comprised code inserted (or patched) into the code of an executable program.

Patch 1: Wide, wide, narrow, narrow. This is used for post-scan image control.

Patch 2: Wide, narrow, narrow, wide. This code assigns “image level 2” to the scanned document.

Patch 3: Wide, narrow, wide, narrow. This code assigns “image level 3” to the scanned document.

Patch 4: Narrow, wide, wide, narrow. Patch 4 is used for post-scan image control.

Patch 5: Narrow, narrow, wide, wide. This Patch Code is also used for image control.

Patch T 6: Narrow, wide, narrow, wide. Also known as the “transfer patch,” this code assigns a predefined image level to the next scanned document.

- ✓ **Jumper cable types**

Jumper wires typically come in three versions: male-to-male, male-to-female and female-to-female. The difference between each is in the end point of the wire. Male ends have a pin protruding and can plug into things, while female ends do not and are used to plug things into

- ✓ **Patch panel management Techniques**

Why Need Patch Panel Cable Management?

The most part is that a patch panel provides a centralized location to manage network connections. When it comes to making a move, add, or change (MAC), the patch panel cable management would effectively reduce the time and cost to perform physical changes at a patch panel in a wiring closet.

1. Rack Mount Enclosure + Fiber Patch Panel

The rack mount enclosure is always loaded with LC, SC, ST, MTP/MPO fiber adapter panel to provide a pathway to connect backbone-to-backbone or backbone-to-horizontal fiber cabling.

2. Rack Mount Enclosure + Fiber Optic Cassette

In addition to mounting with fibre optic patch panel, rack mount enclosure can also hold MTP-8, MTP-12, or MTP-24 fibre cassette to provide the interface between the MTP connector on the trunk and the LC duplex jumpers for quick connection of remote or data centre applications.

3. Blank Rack Mount Modular Panel + Fiber Optic Cassette

The blank modular panel has multiple functions to provide a complete solution for routing network cabling and protecting patch cords.

4. Blank Multimedia Adapter Patch Panel + Cable Management Panel

Blank multimedia adapter patch panel allows customization of installation for multimedia applications requiring integration of fibre patch cables and copper cables.

5. Ethernet Patch Panel + Horizontal Cable Manager with D-rings

Ethernet patch panel includes Cat5e, Cat6, or Cat7 patch panel. They are an ideal method to create a flexible, reliable and tidy cabling system for Ethernet cables.

- **Testing of system connectivity**

A VoIP test is a diagnostic tool that measures the quality of your internet connection and evaluates its ability to support VoIP calls.

It checks various parameters like bandwidth, latency, jitter, packet loss, and MOS score (Mean Opinion Score) to determine the quality of your internet connection.

Points to Remember

- ✓ **Connection of connectors**

Straight -Tip (ST): An older bayonet style connector widely used with multimode fibre.

Subscriber Connector (SC): Sometimes referred to as square connector or standard connector. It is a widely adopted LAN and WAN connector that uses a push-pull mechanism to ensure positive insertion. This connector type is used with multimode and single-mode fibre.

Lucent Connector (LC): Sometimes called a little or local connector, is quickly growing in popularity due to its smaller size. It is used with single-mode fibre and supports multimode fibre.

Application of learning 2.3.

A small office has recently implemented a VoIP phone system. The IT technician is tasked with connecting a new employee's IP phone to the network. As technician you must ensure that the phone is correctly configured, has access to the network, and can make and receive calls.

Learning outcome 2 end assessment

Theoretical assessment

Q1. Choose the correct answer.

i. Which of the following is NOT a key component of a VoIP system?

   A. PBX
   B. Modem
   C. VoIP phone
   D. Media server

ii. What is the primary advantage of using VoIP over traditional phone systems?

   A) Higher call quality
   B) Lower cost
   C) More features
   D) All the above

iii. Which protocol is commonly used for VoIP communication?

   A) HTTP
   B) FTP
   C) SIP
   D) SMTP

iv. What is the most common method of connecting a VoIP phone to a network?

   A. Wireless
   B. Ethernet
   C. Analog
   D. Digital

v. What is the purpose of a Power over Ethernet (PoE) switch in VoIP phone connections?

   A. To amplify the network signal.
   B. To provide power to VoIP phones through the network cable.
   C. To convert analogy signals to digital.
   D. To enhance call quality.

Q2. Answer by True or False

- **i)** VoIP systems require higher bandwidth compared to traditional phone systems.
- **ii)** QoS is essential for optimal VoIP performance.
- **iii)** A VoIP phone can be used with any traditional phone line.
- **iv)** All VoIP phones require a separate power adapter.
- **v)** An RJ-45 connector is used to terminate Ethernet cables.
- **vi)** A VoIP phone can be connected directly to a computer's Ethernet port.

Q3. Match the following VoIP components with their uses:

SN	VOIP components	Uses
1	Codec	A. Ensures quality of VoIP calls by prioritizing voice traffic
2	PBX	B. Handles internal and external phone calls within an organization
3	VoIP Gateway	C. Converts analogy voice signals into digital data
4	SIP	D. Connects an analogy phone system to a VoIP network
5	QoS	E. Signalling protocol for VoIP communication

Practical assessment

XXX organization, a small-to-medium sized business, has decided to implement a VoIP system to replace its traditional analogy phone system. The goal is to reduce communication costs, improve call quality, and enhance overall communication efficiency. You are tasked with deploying the VoIP system equipment and configuring it for basic functionality.

Tasks:

1: Unpack and inspect VoIP system equipment.
2: Install and configure the IP PBX.
3: Connect VoIP phones to the network.
4: Configure VoIP phones for basic operation.
5: Test the VoIP system functionality.
6: Troubleshoot any issues encountered during deployment.

END

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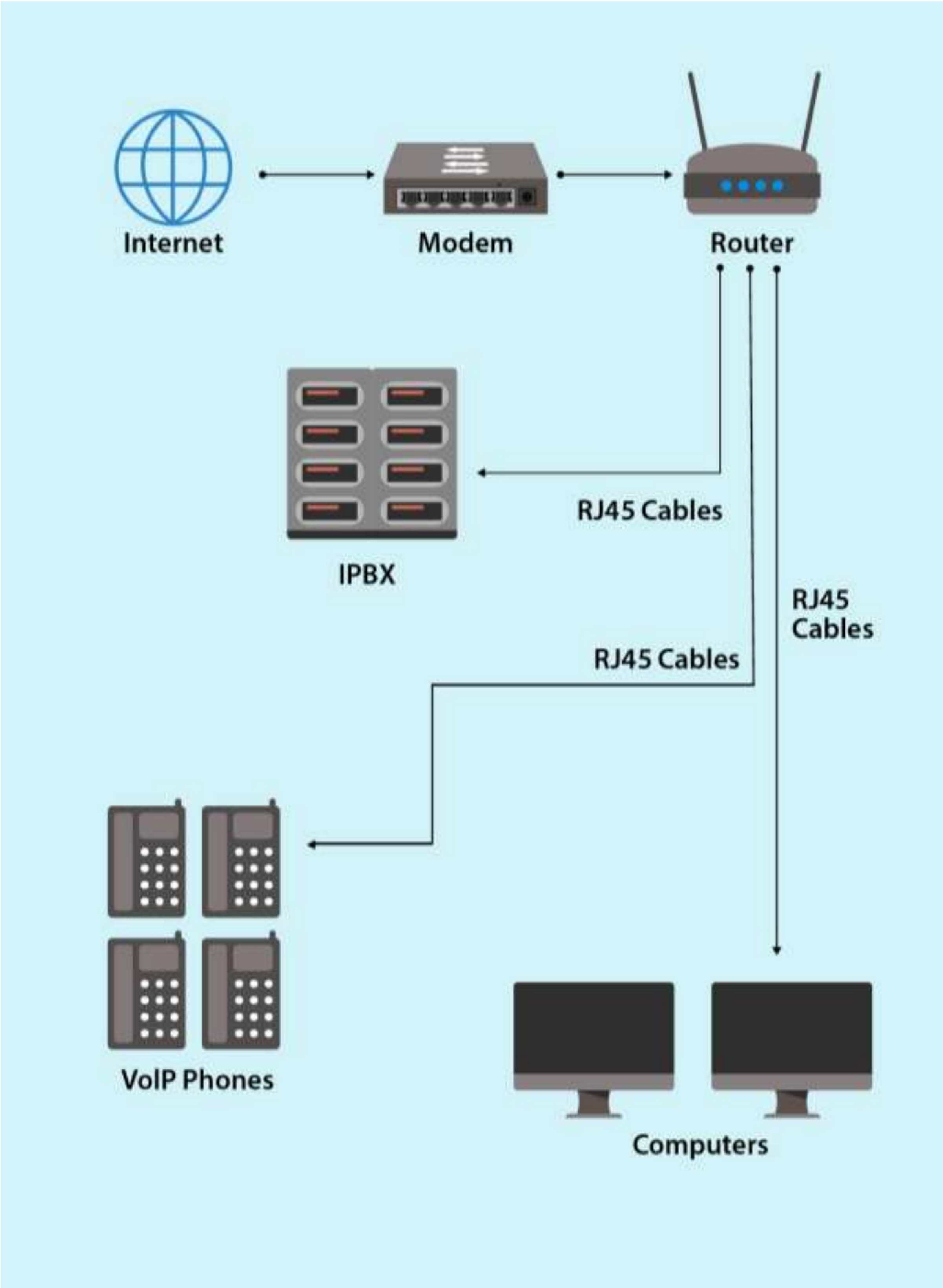

Internet
Modem
Router
IPBX
RJ45 Cables
RJ45 Cables
RJ45
Cables
VoIP Phones
Computers

Indicative contents
3.1. Configuration of Router Based VoIP system. **3.2. Configuration of phone line number** **3.3. Configuration of phone call manager (CME)** **3.4. Configuration of VoIP system Server based (IP PBX/PABX/PBX)** **3.5. Configuration of Calls Routing** **3.6. Testing of VoIP system (Speed, Latency, Jitter, Packet Loss and QoS)**

Key Competencies for Learning Outcome 3: Configure VoIP system

Knowledge	Skills	Attitudes
• Configuration of Router-based VoIP system • Configuration of phone line number. • Configuration of phone call manager (CME) • Configuration of VoIP system Server based (IP PBX/PABX/PBX). • Configuration of calls routing • Description of testing of VoIP system.	• Configurating router-based VoIP System. • Creating of VoIP pool for voice system • Configuration phone line number • Configurating phone call manager (CME) • Configurating VoIP server based (IP PBX/PABX/PBX) • Configurating calls routing. • Testing of VoIP system	• Being attentive • Having discipline, • Being patience, • Being persistence, • Having teamwork • Being Researcher,

Duration: 30 hrs

Learning outcome 3 objectives:

By the end of the learning outcome, the trainees will be able to:

1. Configure properly Router based according to the VoIP system design.
2. Create correctly the VoIP pool according to the VoIP system design.
3. Configure properly phone line number based on VoIP system design.
4. Configure properly Call manager based on VoIP system design.
5. Configure properly VoIP server based (IP PBX/PABX/PBX) based on VoIP system requirement.
6. Configure call routing based on VoIP system requirements.
7. Describe appropriately testing of VoIP system based on the configured VoIP system.
8. Test correctly VoIP system based on the configured VoIP system.

Resources

Equipment	**Tools**	**Materials**
• Channel bank • IP phone • Power backup (UPS) • IP PBX Server/PBX/PABX Server VoIP Router • VoIP Gateway • VoIP Headset • PoE Switch • Computer	• Simulators (Packet tracer, GNS3) • Operating system (Linux)	• Internet bundles

Indicative content 3.1: Configuration of Router Based VoIP system.

Duration: 5 hrs

Theoretical Activity 3.1.1: Configuration of VoIP essential router-based

Tasks:

Task 1: Answer the following questions related to the VoIP essential router-based configurations.

i. What do you understand about the following keyword in router-based configuration?
   a) Interface
   b) Default gateway and
   c) Password

ii. Describe the step of configuring VOIP router interface, default gateway and password?

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to the whole class.

Task 4: In addition, ask questions where necessary.

Task 5: For more clarification, read the key **readings 3.1.1**

Key readings 3.1.1.: Configuration of VoIP essential router-based

- **VoIP essential router-based configurations**

✓ **Interface**

A network interface is the point of interconnection between a computer and a private or public network. A network interface is generally a network interface card (NIC) but does not have to have a physical form. Instead, the network interface can be implemented in software. For example, the loopback interface (127.0.0.1 for IPv4 and :1 for IPv6) is not a physical device but a piece of software simulating a network interface.

✓ **Default gateway**

Is the IP address that serves as the entry point or exit point for traffic leaving or entering a local network. It is a critical part of network configuration because it determines how traffic is routed between the local network and external networks, such as the internet.

- **Password**

A "password" typically refers to one of several types of passwords used to secure and manage the router's settings and access to its configuration interface.

here are several different types of passwords associated with a router:

- **Router Login Password**: This is the most common password used to access the router's configuration settings. It's also known as the "admin password" or "router login password." This password is required when logging in to the router's web-based configuration interface. It's used to prevent unauthorized access and make changes to the router's settings. It's essential to change the default router login password to enhance security.
- **Wireless Network Password (Wi-Fi Password)**: This is a password used to secure your wireless network. It is also commonly referred to as the "Wi-Fi password" or "WPA/WPA2 passphrase." This password is used by devices to connect to the wireless network. It's important to set a strong and unique Wi-Fi password to prevent unauthorized access to your network.
- **Enable Password**: In some routers, there is an "enable password" or "enable secret" password. This password is used to access privileged EXEC mode or administrative mode in the router's command-line interface. It helps protect critical router commands from unauthorized access.
- **Console Password**: Routers often have a console port that can be accessed via a direct physical connection (console cable). A console password is used to secure access to the router's console port, which is typically used for local management of the router.
- **Telnet or SSH Password**: If you access the router remotely over a network using protocols like Telnet or SSH, you may need a password to authenticate and gain access. These passwords are used to secure remote access to the router.
- **Virtual Terminal (VTY) Password**: Routers that allow remote access via Telnet or SSH have virtual terminal lines (VTY lines). A VTY password is used to secure access to these virtual terminal lines.

❖ **Steps of configuring interface, default gate way and password.**

1. **Access the Router**

Access the router's command-line interface through a terminal program or console cable. You'll need to connect to the router's console port using a terminal emulator like PuTTY or a similar program.

2. **Enter Configuration Mode**:

Once you have access to the router, enter the configuration mode.

```
enable
configure terminal
```

3. **Configure the Interface**:

In this example, we'll configure a FastEthernet interface with the IP address 192.168.1.1 and a subnet mask of 255.255.255.0. Adjust the interface name, IP address, and subnet mask to match your network setup:

```
interface FastEthernet0/0
ip address 192.168.1.1 255.255.255.0
no shutdown
exit
```

4. **Set the Default Gateway**

To set the default gateway for the router, use the following command.

```
ip route 0.0.0.0 0.0.0.0 [gateway IP]
```

Replace [gateway IP] with the IP address of your upstream router or gateway.

5. **Configure Passwords**

In this example, we'll set a password for the console line and the VTY (Virtual Terminal) lines. You should replace [console-password] and [vtty-password] with your chosen passwords.

- Console Line Password

```
line con 0
password [console-password]
login
```

- **VTY Line Password (for remote access)**

```
line vty 0 15
password [vtty-password]
login
```

 **Practical Activity 3.1.2: Configurating VoIP essential router-based**

Tasks:

Task 1: Referring to the key reading (3.1.2) you are requested to perform the given task. The task should be done individually.

Configure the router using Faster Ethernet interface has following address 192.168.1.1, default gateway 192.168.1.1 and console password "VOIP1" by referring to the key reading 3.1.2.

Task 2: List out procedures to be used to perform the given tasks.
Task 3: Trainees follow the instruction
Task 4: Referring to procedures provided on task 3.1.2, Perform the given tasks.
Task 5: Present your work to the trainer and whole class.
Task 6: Ask clarification where necessary
Task 7: Perform the task provided in application of learning 3.1. and read **key reading 3.1.2.**

 **Key readings: 3. 1.1: Configurating VoIP essential router-based**

Steps followed to configure VoIP router-based are:

Step 1: Open packet tracer as simulation software tool
Step 2: chose the router you will configure.
Step 3: open router in command line interface
Step4: type the following command in router.

```
Router>enable
Router#configure terminal.
Router(config)#interface fastEthernet 0/0
Router(config-if)#ip address 192.168.1.1 255.255.255.0
Router(config-if)#exit
Router(config)#line console 0
Router(config-line)#password VOIP1
Router(config-line)#Login
Router(config-line)#exit
Router(config)#exit
Router#
%SYS-5-CONFIG_I: Configured from console by console
Router#write memory
Building configuration...
[OK]
```

Theoretical Activity 3.1.3: Creation of VoIP Pool for Voice system.

Tasks:

Task 1: Answer the following questions related to the Creation VoIP Pool for Voice system

i. What is pool name?
ii. What is the purpose of pool in the way of creating VoIP Pool for Voice system?
iii. Why is voice option (150 option) needed?
iv. Write down all steps configuring pool name, pool address and voice option?

Task 2: Provide the answer for the asked questions and write them on papers and present the findings/answers to trainer and the whole class.

Task 3: In addition, ask questions where necessary.

Task 4: For more clarification, read the **key readings 3.1.3.**

Key readings 3.1.3: Creation of VoIP Pool for Voice system

✓ **Pool Name**

The "Pool name" is a user-defined identifier for the VoIP pool. It's a friendly name used to distinguish this pool from others and make it easier to manage. For example, you might give your VoIP pool a name like "Voice Devices" or "VoIP Phones."

✓ **Pool Address (Network Address and Subnet Mask)**

The "Pool address" consists of two components: the network address and the subnet mask. Together, they define the range of IP addresses that will be allocated to VoIP devices. This range is used to ensure that VoIP devices receive IP addresses within a specific subnet. For example, if you specify a network address of 192.168.1.0 and a subnet mask of 255.255.255.0, you are defining a pool of IP addresses from 192.168.1.1 to 192.168.1.254 within the 192.168.1.0/24 subnet.

✓ **Voice Option (150 option)**

Is the number of IP addresses you want to allocate within the VoIP pool. In your example, "150 option" suggests that you want to allocate 150 IP addresses to VoIP devices within the specified subnet. This is the number of addresses available for VoIP phones to use. The specific number of IP addresses allocated will depend on the size and needs of your VoIP deployment. You can adjust this number as required.

- **Steps configuring pool name, pool address and voice option.**

Step 1: Access the Configuration Interface

Access the configuration interface of your network equipment. This could be a router, DHCP server, or other networking devices.

Step 2: Define the Pool Name

In the configuration interface, create a pool name to identify your VoIP pool. The exact steps for this may vary depending on your equipment. Look for a section or option related to DHCP or IP address allocation.

- Example (using Cisco router commands):

configure terminal

ip dhcp pool VoiceDevices

Step 3: Specify the Pool Address (Network Address and Subnet Mask)

3. Define the network address and subnet mask for your VoIP pool. This establishes the range of IP addresses that will be allocated to VoIP devices.
   - Example (using Cisco router commands):

   network 192.168.1.0 255.255.255.0

3. In this example, we've specified a network address of 192.168.1.0 and a subnet mask of 255.255.255.0, creating a /24 subnet.

Step 4: Voice Option (Allocate IP Addresses)

The "voice option" in your context appears to refer to the number of IP addresses you want to allocate to VoIP devices. You can specify the range of IP addresses to allocate within the subnet. The number of IP addresses allocated will depend on the specific requirements of your VoIP deployment.

Example (using Cisco router commands)

ip dhcp pool VoiceDevices

network 192.168.1.0 255.255.255.0

address range 192.168.1.10 192.168.1.59 # Allocating 50 IP addresses (adjust as needed)

In this example, we've allocated a range of 50 IP addresses within the subnet for VoIP devices.

Step 5: Save and Apply Configuration

After configuring the pool name, pool address, and voice option, make sure to save and apply the configuration changes in your networking equipment.

- **Example (using Cisco router commands):**

End write memory

Step 6: Test and Monitor

Test the configuration to ensure that VoIP devices receive the allocated IP addresses from the VoIP pool. Monitor your VoIP system's performance to ensure that it's working as expected.

Practical Activity 3.1.4: Creating VoIP Pool for Voice system.

Tasks:

Task 1: Referring to key reading (3.1.4) you are requested to perform the given task. The task should be done individually.

In configuration VoIP router we have two ways of assigning address, DHCP and Static assignment, you are the one who studied the way of creating VOIP Pool for Voice system using packet tracer simulation create VoIP Pool name and pool address.

Task 2: List out procedures to be used to perform the given tasks (3.1.2)

Task 3: Trainee follow the instruction given by trainer.

Task 4: Referring to procedures provided on task 2, Perform the given tasks and present your work to the trainer and whole class.

Task 5: Ask clarification where necessary

Task 6: Read **key reading 3.1.4** and perform the task provided in application of learning 3.1.

Key readings: 3.1.4: Creating VoIP Pool for Voice system.

A VoIP (Voice over Internet Protocol) pool for a voice system refers to a collection of virtual or physical communication resources, such as servers, lines, and channels, dedicated to managing and facilitating voice communications over the Internet. These pools are designed to efficiently handle voice traffic, ensuring reliable and high-quality communication for users.

By allocating and distributing resources dynamically, VoIP pools optimize the utilization of available bandwidth and infrastructure, enabling scalable and cost-effective voice systems. VoIP technology transforms analog voice signals into digital data packets for transmission over

IP networks, offering businesses and individuals a flexible and cost-efficient solution for voice communication that can integrate seamlessly with other data services.

The VoIP pool's effectiveness relies on sophisticated routing algorithms and network management protocols to prioritize and deliver voice traffic in a timely and efficient manner.

The following process of creating pool name and pool address in VoIP system using Packet tracer simulation.

```
Router(config-if)#
%LINK-5-CHANGED: Interface FastEthernet0/0, changed s

Router(config-if)#
Router(config-if)#exit
Router(config)#
Router(config)#
Router(config)#ip dh
Router(config)#ip dhcp po
Router(config)#ip dhcp pool voip
Router(dhcp-config)#net
Router(dhcp-config)#network 192.168.10.0 255.255.255.
Router(dhcp-config)#defau
Router(dhcp-config)#default-router 192.168.10.1
Router(dhcp-config)#
```

Points to Remember

- A network interface is the point of interconnection between a computer and a private or public network. A network interface is generally a network interface card (NIC) but does not have to have a physical form. Instead, the network interface can be implemented in software.
- Is the IP address that serves as the entry point or exit point for traffic leaving or entering a local network. It is a critical part of network configuration because it determines how traffic is routed between the local network and external networks, such as the internet.
- A "password" typically refers to one of several types of passwords used to secure and manage the router's settings and access to its configuration interface.

- The "**Pool name"** is a user-defined identifier for the VoIP pool. It's a friendly name used to distinguish this pool from others and make it easier to manage.
- **The "Pool address"** consists of two components: the network address and the subnet mask. Together, they define the range of IP addresses that will be allocated to VoIP devices.

- **150 option** Is the number of IP addresses you want to allocate within the VoIP pool. In your example, "150 option" suggests that you want to allocate 150 IP addresses to VoIP devices within the specified subnet.
- **A VoIP (**Voice over Internet Protocol) pool for a voice system refers to a collection of virtual or physical communication resources, such as servers, lines, and channels, dedicated to managing and facilitating voice communications over the Internet. These pools are designed to efficiently handle voice traffic, ensuring reliable and high-quality communication for users.

Application of learning 3.1.

In KIMARANZARA restaurant use static assignment for assigning new endpoint (telephone) to their VoIP system, this way makes hardworking and disturbance in the network. This restaurant needs your assistant where you will change the way of new end point (telephone).

Indicative content 3.2: Configuration of Phone Line Number

Duration: 5 hrs

Theoretical Activity 3.2.1: Configuration Phone directory for IP Phone and ephones

Tasks:

Task 1: In small groups, you are requested to answer the following questions related to the configuration Phone directory for IP Phone and ephones

i. What is phone directory for IP Phone?
ii. What is ephone?
iii. Describe the feature of phone directory for IP phone in VOIP system.
iv. Elaborate the step of configuring Phone directory for IP phone and ephones?

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to the whole class and ask questions where necessary.

Task 4: For more clarification, read the key **readings 3.2.1**

Key readings 3.2.1.: Configuration Phone directory for IP Phone and ephones

- **Phone directory for IP Phone**

IP phones is a digital database or feature that stores and provides access to a list of contact information, typically in the form of phone numbers and associated details, for the users of IP phones within the organization.

This directory helps users quickly and easily find and call their colleagues, clients, or other contacts using their IP phones.

A phone directory for IP phones in a VoIP system is a feature that:

- ✓ **Stores Contact Information**: It maintains a digital record of contact information, which may include names, phone numbers, email addresses, and other relevant details of individuals or entities associated with the organization.
- ✓ **Provides Search and Access**: Users of IP phones can search, browse, or access this directory directly from their phones, usually through a user-friendly interface on the phone's display.
- ✓ **Facilitates Dialing**: Once a user selects a contact from the directory, the IP phone can initiate a call to that contact by automatically dialing the associated phone number. This simplifies the process of making calls and reduces the need for manual entry of numbers.

- ✓ **Integration**: The phone directory may integrate with other features or services within the VoIP system, such as presence information (indicating whether a contact is available or busy), call history, and voicemail.
- ✓ **Centralized Management**: In larger organizations, the phone directory is often centrally managed, ensuring consistency and accuracy of contact information across the organization.
- ✓ **Security and Access Control**: Access to the phone directory may be controlled to protect sensitive contact information. Typically, authorized users can access and edit the directory, while other users have read-only access.
- ✓ **Updates and Synchronization**: The directory may support automatic updates or synchronization with other databases or contact management systems to ensure that the information is always up-to-date.
- ✓ **Personal and Corporate Contacts**: Some phone directories allow users to maintain their personal contacts alongside corporate contacts, making it easy to access both work-related and personal contacts from the same IP phone.
- ✓ **Customization**: Organizations can often customize the appearance and layout of the directory, as well as the fields associated with each contact, to suit their specific needs.

- **Ephones**

An "ephone" stands for "electronic phone" or "Ethernet phone." Ephone is a term used in Cisco's Unified Communications solutions and Cisco's IP Telephony systems, specifically the Cisco Unified Communications Manager (CUCM) and Cisco Unified Communications Express (CUCME). An ephone is a virtual representation of an IP phone or endpoint within a VoIP system, and it is associated with a physical device or softphone that users use to make and receive phone calls over an IP network.

How to make configuration of Phone directory for IP Phone and ephones?

- ❖ **For Cisco Unified Communications with ephones (using Cisco CME)**:

1. **Access Router Configuration Mode**: Log in to the router that is running Cisco Unified Communications Manager Express (CME).
2. **Define Phone Directory Entries**: Use the ephone-dn command to define individual phone directory entries. These entries will include the name and extension number.

```
 ephone-dn 1
 number 1001
 name John Doe
!
ephone-dn 2
 number 1002
```

name Jane Smith
!

3. **Create a Phone Directory List**:

Define a directory list that includes the ephone-dn entries. The directory list can have a name like "InternalDirectory."

telephony-service
directory entry 1 1001 name John Doe
directory entry 2 1002 name Jane Smith

4. **Enable Directory Lookup on IP Phones**:

On each IP phone (ephone), enable the directory lookup feature so that users can access the phone directory.

ephone 1
button 1:1
directory entry 1 1001
directory entry 2 1002

This assigns directory entries to specific buttons on the IP phones.

- **For Other VoIP Systems (non-Cisco)**

1. **Access the VoIP System Configuration**: Access the configuration interface of your VoIP system, which could be a web-based interface or a dedicated management platform.
2. **Create a Directory or Contacts List**: In the configuration interface, create a list of contacts or directory entries. Each entry should include the contact's name, phone number, and other relevant information.
3. **Import or Manually Add Contacts**: Depending on your VoIP system, you may have the option to import contacts from a file or manually add them one by one.
4. **Assign Directory Access to IP Phones**: Configure the IP phones to access the directory or contacts list. The steps for this may vary depending on the specific VoIP system.

Test and Verify: Test the configuration to ensure that users can access and use the phone directory on their IP phones.

Points to Remember

IP Phone: A physical device that uses internet protocol (IP) to make and receive calls. Examples include Cisco IP Phones, Polycom phones, etc.

ePhone: A software-based representation of a phone line or directory number within a VoIP system. It's a configuration entity, not a physical device.

Application of learning 3.2:

In VOIP system each end user in own number for help her/him to make call, this number provided by who responsible of VOIP System configuration. So, the company XYXY need to provide number to their employees you are the one who responsible make these activities. The task is configuring phone line number to all employees.

Indicative content 3.3: Configuration of Phone Call Manager (CME)

Duration: 5 hrs

Theoretical Activity 3.3.1: Configuration of phone call manager (CME)

Tasks:

Task 1: Answer the following questions related to the configuration of Telephone services, Maximum number of ephones, Maximum number of directory (dn) and Auto assignment of line numbers.

i. Elaborate the at least 5 telephone services that can provided by VoIP systems?
ii. Distinguish factors that influence the maximum number of ePhones in a VoIP system?
iii. Mention Factors that can influence the maximum number of directory numbers?
iv. Identify the Benefits of Auto Assignment of Line Numbers in VoIP Systems
v. Determine the procedure of configuring Telephone services, Maximum number of ephones, and Maximum number of directories in VoIP system.

Task 2: Provide the answer for the asked questions and write them on papers and present the findings/answers to the whole class

Task 3: In addition, ask questions where necessary.

Task 4: For more clarification, read the **key readings 3.3.1.**

Key readings 3.3.1: Configuration of phone call manager (CME)

- **Telephone services**

Telephone services in a Voice over Internet Protocol (VoIP) system refer to the functionality that allows users to make and receive phone calls over the internet using VoIP technology. VoIP is a technology that converts voice and other multimedia content into digital data packets and transmits them over the internet or private data networks. It offers several telephone services that are like traditional landline or mobile phone services, but with some key differences and advantages.

Some common telephone services provided by VoIP systems.

✓ **Voice Calling**: VoIP systems enable voice calls, allowing users to make and receive phone calls to and from other VoIP users, as well as traditional landline and mobile phones. The voice quality can be high if the internet connection is stable and has sufficient bandwidth.

- ✓ **Video Calling**: Many VoIP services support video calls, which allow users to see and interact with each other through live video feeds. Video conferencing is a common use case for business communications.
- ✓ **Messaging**: VoIP systems often include text-based messaging features, such as instant messaging or chat, which can be used for real-time communication or asynchronous messaging.
- ✓ **Voicemail**: VoIP systems usually provide voicemail services, allowing callers to leave recorded messages when the recipient is unavailable. Users can access their voicemail messages through various devices, including computers and mobile apps.
- ✓ **Call Forwarding**: VoIP systems often offer call forwarding options that allow users to redirect incoming calls to another phone number or device, ensuring that calls are not missed.
- ✓ **Call Waiting and Call Holding**: Like traditional phone services, VoIP systems typically support call waiting (notifying users of incoming calls while on another call) and call holding (temporarily placing a call on hold).
- ✓ **Call Transfer**: Users can transfer calls to another party or extension, which is useful for businesses and call centers to route calls efficiently.
- ✓ **Caller ID**: VoIP services often display caller ID information, showing the name and phone number of the incoming caller.
- ✓ **Conference Calling**: VoIP allows for multi-party conference calls, making it easy to set up and participate in audio and video conferences with multiple participants.
- ✓ **Directory Services**: VoIP systems may provide directory services for internal and external contacts, making it easy to find and connect with other users.
- ✓ **Emergency Services**: Many VoIP providers offer access to emergency services (e.g., 911) and provide location information to first responders, but this may vary depending on the service and location.

- **Maximum number of ephones**

The maximum number of ePhones (electronic phones) that can be supported in a Voice over Internet Protocol (VoIP) system varies depending on several factors, including the specific VoIP platform or solution being used, the hardware and network infrastructure in place, and the capacity of the VoIP server or PBX (Private Branch Exchange) system. There isn't a fixed maximum number that applies universally, and it can be tailored to the requirements of the organization or deployment.

Factors that influence the maximum number of ePhones in a VoIP system

- ✓ **VoIP Server Capacity**: The capacity of the VoIP server or PBX is a critical factor. Higher-capacity servers can handle more concurrent phone connections. The exact number of ePhones a server can support will depend on the server's processing power, memory, and network capabilities.

- ✓ **Network Bandwidth**: The available network bandwidth is essential. VoIP calls require a certain amount of bandwidth for voice data transmission. The more ePhones in use simultaneously, the more bandwidth is needed. Organizations must ensure that their network infrastructure can handle the expected call volume.
- ✓ **Codec Selection**: The choice of audio codecs can impact the number of ePhones a VoIP system can support. Some codecs use less bandwidth but may sacrifice audio quality, while others offer higher quality at the cost of more bandwidth. The codec used can affect the total number of simultaneous calls.
- ✓ **Hardware and Resources**: The hardware used for ePhones, such as IP phones or softphones, will also influence the system's capacity. More powerful phones or computers may support a greater number of active calls.
- ✓ **Licensing and Software Limits**: The specific VoIP software and licenses in use may impose limits on the number of ePhones supported. Some VoIP platforms charge per user or device, which can impact scalability.
- ✓ **Network Quality of Service (QoS)**: Implementing Quality of Service in the network can help ensure that VoIP traffic is prioritized, reducing the risk of call quality degradation when many ePhones are in use.
- ✓ **Redundancy and Failover**: Implementing redundancy and failover mechanisms can affect the number of ePhones supported since additional resources may be needed for backup systems.
- ✓ **Network Topology**: The network topology and the architecture of the VoIP deployment can affect scalability. For example, a distributed system with multiple servers may be able to support more ePhones compared to a single server setup.

- **Maximum number of directory (dn)**

The maximum number of directory numbers (DN) in a Voice over Internet Protocol (VoIP) system, including a VoIP server or PBX (Private Branch Exchange), can vary widely depending on the specific VoIP platform, hardware, and licensing arrangements. There is no universal fixed limit for the number of directory numbers in a VoIP system, as it can be customized to meet the needs of the organization or deployment.

Factors that can influence the maximum number of directories numbers

- ✓ **VoIP Server or PBX Capacity**: The capacity of the VoIP server or PBX plays a significant role. More powerful servers can handle a larger number of directory numbers. The server's processing power, memory, and network capabilities are important considerations.
- ✓ **Licensing and Software Limits**: The specific VoIP software and licensing arrangements in use can impose limitations. Some VoIP platforms charge based on the number of directory numbers or users, so licensing can directly impact scalability.

- ✓ **Hardware Resources**: The hardware used for the VoIP system, including IP phones and the server, can influence the number of directory numbers supported. More powerful hardware may be able to handle a larger number of DNs.
- ✓ **Network Bandwidth:** The network's available bandwidth can be a limiting factor, especially if many directory numbers are in use simultaneously. VoIP calls require a certain amount of bandwidth for voice data transmission.
- ✓ **Call Handling Capacity:** The VoIP system's ability to handle concurrent calls, call routing, and other features can also affect the maximum number of directory numbers.
- ✓ **Redundancy and Failover:** If redundancy and failover mechanisms are in place for the VoIP system, additional resources may be needed, impacting the number of directory numbers that can be supported.
- ✓ **Network Quality of Service (QoS):** Implementing Quality of Service in the network can help ensure that VoIP traffic is prioritized, which is crucial for maintaining call quality when multiple directory numbers are in use.
- ✓ **Network Topology:** The network architecture and the VoIP system's design can influence scalability. A distributed system with multiple servers may have a different capacity compared to a single-server setup.

- **Auto assignment of line numbers**

Auto assignment of line numbers in a VoIP (Voice over Internet Protocol) system refers to the automated process of assigning phone line numbers or directory numbers (DNs) to individual users or devices within the VoIP system. This functionality is particularly useful in larger organizations or businesses where managing many phone lines can be complex and time-consuming.

How auto assignment of line numbers typically works in a VoIP system.

1. **User or Device Registration**: When a new user or device is added to the VoIP system, such as an IP phone, softphone application, or other communication endpoint, the system needs to assign a unique line number or directory number for that user or device.
2. **Auto Assignment Logic**: The VoIP system is configured with a set of rules or logic for auto-assigning line numbers. These rules may include specific patterns, ranges, or algorithms for generating line numbers. For example, line numbers might be assigned based on department, location, or user role.
3. **Available Line Numbers**: The system maintains a database of available line numbers or DNs that have not been assigned to other users or devices. These can be preconfigured or dynamically generated based on the auto assignment rules.
4. **Automatic Assignment**: When a new user or device is registered, the VoIP system automatically selects an available line number according to the predefined rules.

This process happens without manual intervention, making it efficient and error-free.

5. **Configuration and Provisioning**: Once the line number is auto assigned, the system configures the user or device with the assigned number and associates it with the necessary features and services, such as voicemail, call forwarding, and more.
6. **User Accessibility**: Users can then access their assigned line numbers for making and receiving calls, as well as accessing other communication services offered by the VoIP system.

Benefits of Auto Assignment of Line Numbers in VoIP Systems:

1. **Efficiency**: Auto assignment streamlines the process of setting up new users or devices, reducing the administrative workload and minimizing the potential for errors in manual assignments.
2. **Consistency**: It ensures that line numbers are assigned according to a consistent and predefined logic, making it easier to manage and understand the numbering scheme.
3. **Scalability**: As an organization grows, it can easily accommodate new users and devices without the need for manual configuration, making it scalable.
4. **Resource Optimization**: By dynamically selecting available line numbers, it optimizes the allocation of resources within the VoIP system.
5. **Flexibility**: Auto assignment rules can be adjusted or customized to meet the organization's specific needs and policies.

Steps of configuring Telephone services, Maximum number of ephones, Maximum number of directory (dn) and Auto assignment of line numbers.

Step1: Access CME Configuration Mode:

Access the router running CME and enter the configuration mode.

```
configure terminal
```

Step2: **Configure Telephone Services**

Define the telephone service settings, including the phone service name and the XML file that provides the phone's display information. This information is typically displayed on the IP phone's screen.

```
telephony-service
 service phone displayOnStartup 1
 service phone name CiscoIPPhone
 service phone XMLDefault.cnf.xml
```

Step 3: Define Maximum Number of Ephones

Specify the maximum number of ephones (IP phones) that can be configured in the system. Adjust this number based on your organization's needs.

```
telephony-service
 max-ephones [maximum number]
 max-dn [maximum directory numbers]
```

Step 4: Configure Auto-Assignment of Line Numbers:

Enable the automatic assignment of line numbers for ephones. This allows the system to automatically allocate extension numbers as ephones are added.

```
telephony-service
 auto-reg-ephone
```

This setting enables phones to auto-register and receive line numbers sequentially.

Step 5: Define Ephones and DNs

Create ephone-dn and ephone entries for each phone. This step specifies the directory number, name, and associated ephone-dn.

Example for an ephone-dn

```
ephone-dn [number]
 number [directory number]
 name [name]
```

Example for an ephone:

```
ephone [phone number]
 mac-address [MAC address]
 type [phone type]
 button [button number]:[ephone-dn number]
```

Step 6: **Testing and Verification**:

Test the configuration by connecting IP phones and verifying that they register and work correctly. You can check the status of ephones and DNs with the show ephone and show ephone-dn commands.

Step 7: **Save and Exit**:

Save the configuration and exit the configuration mode.

```
end
write memory
```

Points to Remember

- ✓ Telephone services in a Voice over Internet Protocol (VoIP) system refer to the functionality that allows users to make and receive phone calls over the internet using VoIP technology.
- ✓ Auto assignment of line numbers in a VoIP (Voice over Internet Protocol) system refers to the automated process of assigning phone line numbers or directory numbers (DNs) to individual users or devices within the VoIP system. This functionality is particularly useful in larger organizations or businesses where managing many phone lines can be complex and time-consuming.

Application of learning 3.3:

Using router in VoIP system make configuration of telephone services, maximum number of ephone, Maximum number of directory (dn) and Auto assignment of line number for making sure you system serve all connected users.

Indicative content 3.4: Configuration of VoIP System Server Based (IP PBX/PABX/PBX)

Duration: 10 hrs

Theoretical Activity 3.4.1: Configuration VoIP system Server based (IP PBX/PABX/PBX)

Tasks:

Task 1: Answer the following questions related to the Configuration VoIP system Server based (IP PBX/PABX/PBX)

i. Elaborate all steps of configuring VOIP System server based (IP PBX/PABX/PBX).
ii. Differentiate common call routing methods.
iii. Identify at least 3 call routing protocol using in VOIP System configuration.

Task 2: Provide the answer for the asked questions and write them on papers and ask them to present the findings/answers to trainer and whole class

Task 3: In addition, ask questions where necessary.

Task 4: For more clarification, read the **key readings 3.4.1**

Key readings 3.4.1: Configuration VoIP system Server based (IP PBX/PABX/PBX)

- **Steps of Configuration VoIP system Server based (IP PBX/PABX/PBX)**

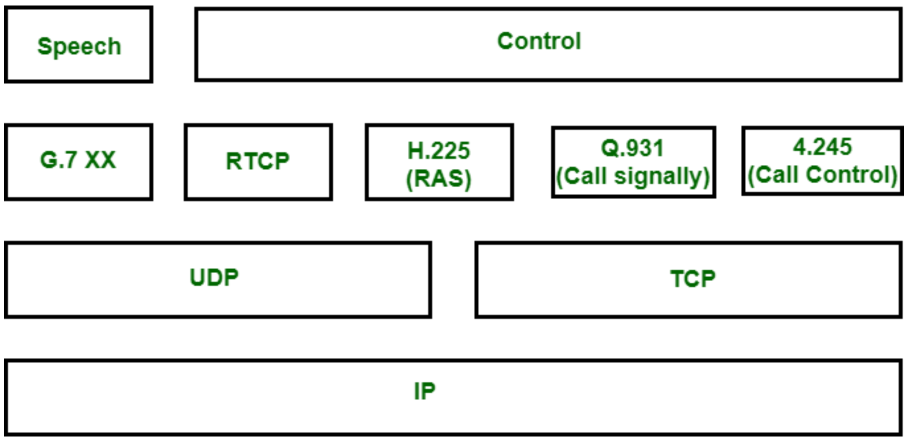
Step1. Installation of Server Operating System: Install and configure the server operating system that will host your VoIP system software. Common choices include Linux-based distributions or Windows Server.

Step2. VoIP Users: Set up user accounts for VoIP system access. Configure usernames, passwords, and access permissions.

Step3. Voice Frequency: Define the voice frequency settings, including the codec used for voice compression and decompression.

Step4. Frame and Voice Packets: Configure voice packetization settings, such as the frame size and the number of voice packets per second.

Step5. Layered Model: Implement a layered model for your VoIP system, which may include components for signaling (e.g., SIP or H.323) and media (e.g., RTP).



Internet protocol Layer in the Model

Network Layer (Layer 3) IP is the foundation of the internet and is responsible for routing data packets between devices on different networks. In the context of VoIP, it is used to route voice and data packets between VoIP devices.

Used datagram protocol (UDP) Layer in the Model

Transport Layer (Layer 4) UDP is a connectionless and lightweight transport protocol used in VoIP for its low overhead and speed. Unlike TCP, which is connection-oriented, UDP does not provide error-checking and retransmission mechanisms. However, in real-time applications like VoIP, where low latency is critical, UDP is preferred because it offers minimal delay.

- **Real time transport protocol (RTP) Layer in the Model**

Application Layer (Layer 7) : RTP is designed for real-time communication and multimedia data. It is used to transmit audio and video streams over IP networks efficiently. RTP provides mechanisms for timestamping, sequencing, and delivering real-time data in the correct order. It also supports the identification of different media types within a single session.

Step6. Inter PBX Circuits: Set up inter-PBX circuits if your organization has multiple PBXs that need to communicate with each other. This may involve configuring trunk lines and gateways.

Step7. Switchboard Facilities: Configure switchboard or receptionist features, allowing for call routing, transfer, and handling of incoming calls.

Step8. Extension Facilities: Configure and manage extension facilities, including assigning extension numbers to users, setting up voicemail, and defining call forwarding rules.

Step9. Public Exchange Services: Establish connections to public exchange services, such as PSTN (Public Switched Telephone Network) or SIP trunk providers, to enable communication with external phone numbers.

Step10: Alarms: Configure alarm notifications to alert administrators about system issues, including hardware, network, or service failures.

Step11: Fuse Alarm and Mains Fail: Implement alarm features to monitor power supply and mains failure, as well as other environmental factors.

Step12: Tones: Set up system tones, including dial tones, ringback tones, busy tones, and error tones for user interaction.

Step13: Numbering: Define numbering plans for internal and external calls, including dialing patterns for local and long-distance calls.

Step14: Local Calls: - Configure local call routing within the organization, allowing users to make calls to other extensions within the system.

Step15: Outgoing Exchange Line/Inter PBX Calls: - Configure outgoing call routing, including routing calls to external lines (PSTN or SIP trunk) and inter-PBX calls if needed.

Step 16: Exchange Line Relay Sets: - Define relay sets for exchange lines, allowing the system to connect to external lines.

Step 17: Interactive Voice Response (IVR) is a technology that allows automated interaction with callers using voice and touch-tone keypad inputs. IVR systems are commonly used in customer service, call centers, and other phone-based services to provide self-service options and route calls efficiently.

Step 18: Call routing methods are techniques used in telephony and call center systems to direct incoming calls to the appropriate destination, such as agents, departments, or locations. These methods determine how calls are distributed, and they can impact factors like fairness, efficiency, and load balancing.

Common call routing methods:

- **Round Robin Call Routing**: In the round-robin method, incoming calls are distributed sequentially, one after the other, to available agents or destinations in a cyclical fashion. This method ensures that each agent or destination receives roughly the same number of calls, promoting fairness and even distribution.
- **Regular Call Routing**: Regular call routing is a straightforward method where calls are routed to available agents or destinations in a linear order. The first available agent receives the next call, and so on. This method does not consider agent skills or other factors, making it simple but not necessarily the most efficient.
- **Uniform Call Routing**: Uniform call routing aims to distribute calls evenly among agents, ensuring that all agents have an equal chance of receiving calls. Unlike regular routing, uniform routing may consider factors like agent availability and skills to achieve a balanced distribution.
- **Simultaneous Call Routing**: In simultaneous call routing, all available agents or destinations receive calls at the same time. Calls are distributed in parallel, and the first agent to answer a call gets the connection. This method is suitable for situations where the fastest response is crucial.
- **Weighted Call Routing**: Weighted call routing assigns different priorities or weights to agents or destinations based on various factors, such as skills, performance, or business rules. Calls are then routed to the highest-priority agents first. This method allows for more intelligent call distribution based on specific criteria.

Step 19: FollowMe: Set up FollowMe features, which allow users to forward calls to other devices or phone numbers when they are not available at their desk.

Step 20: Customer Relationship Management (CRM) Software: Integrate your VoIP system with CRM software to provide call center functionality, call logging, and call history management.

Step 21: Testing and Verification: Thoroughly test and verify the entire configuration to ensure that calls are properly routed, and all system features are functioning as expected.

step22: Ongoing Maintenance and Monitoring: - Implement ongoing maintenance and monitoring procedures to ensure the reliability and performance of the VoIP system.

Practical Activity 3.4.2: Configurating VoIP system Server based (IP PBX/PABX/PBX)

Task:

Task 1: Referring to key reading 3.4.2) you are requested to perform the given task. The task should be done individually.

In configuring VoIP System server (IP PBX/PABX/PBX) there are two option of performing this task, one you can use PBX as electronic devices and also you can you VOIP system server software (Asterisk, Free Switch, Open SIPs, 3CXphone System, Kamaillio, SIP foundry, Free PBX, Elastix, PBX in a flash, Xarcom ltd, Issabel, CallHippo, Sangoma, Digium Shore Tel, Fusin PBX Vital PBX). Using one of above PBX Server software perform VOIP System server.

Task 2: List out procedures and formulas to be used to perform the given tasks.

Task 3: Trainees follow the instructions.

Task 4: Referring to procedures and formulas provided on task 2, Perform the given tasks and present your work to the trainer and whole class.

Task 5: Ask for clarification where necessary.

Task 6: Read **key reading 3.4.2** and perform the task provided in application of learning 3.4.

Key readings 3.4.2: Configurating VoIP system Server based (IP PBX/PABX/PBX)

Elastix is an open source unified communications server software that brings together: IP PBX, Email, IM and Faxing. The Elastix functionality is based on open source projects including Asterisk, HylaFAX, Openfire and Postfix.

Step1. Preparing hardware and software:

1.1 Hardware

First you shall prepare the following items: A PC with an empty HD (what we use herein is SAMSUNG, ATA/133 HDD 80GB), a Synway TEJ-1A/PCI board and a Synway FXM3201P board with one trunk module (CH1 and CH2) and one station module (CH3 and CH4).

You can install the Synway AST series boards either before or after the installation of the Elastix system. Here we install the AST boards first and then install the Elastix system.

All hardware manuals for the AST series boards can be downloaded from the following page.

http://www.synway.net/Support/Resources.aspx

1.2 Software

Make sure you have these software: Elastix 1.5.2, dahdi 2.1.0.4+2.1.0.2 and SynAst-1.5.0.0.

Elastix 1.5.2, about 618MB in size, can be downloaded from:

http://downloads.sourceforge.net/project/elastix/Elastix%20PBX%20Appliance%20Software/1.5.2 /Elastix-1.5.2-stable-i386-bin-31mar2009.iso

Then burn the downloaded driver into a CD.

dahdi-linux-complete-2.1.0.4+2.1.0.2, about 1.8MB in size, can be downloaded from:

http://downloads.asterisk.org/pub/telephony/dahdi-linux-complete/releases/dahdi-linux-complete-2.1.0.4+2.1.0.2.tar.gz

SynAst-1.5.0.0, about 8.62MB in size, can be downloaded from:

http://www.synway.net/Download/Driver/Asterisk/AST1500/SynAST-1.5.0.0_en.tar.gz

Patch for Elastix 1.5.2, provided by Synway, can be downloaded from:

ftp://temp:DOWN@synway.net/AST_Driver/Patch/dahdi-patch-for_elastix1.5.2.rar

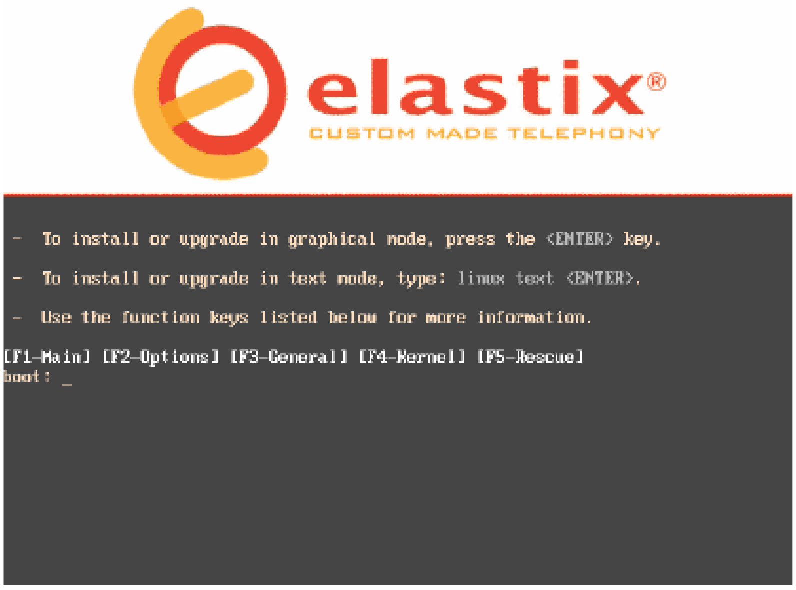
Step 2: Installation of Elastix System

2.1: Set the guide mode

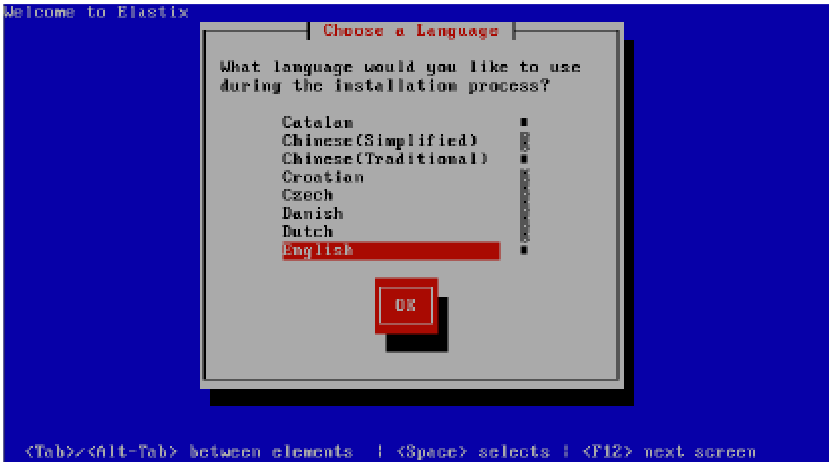
Set BIOS to boot from CD-ROM. Put the CD of Elastix system burned already into CD-ROM and start the PC.

2.2: Install Elastix

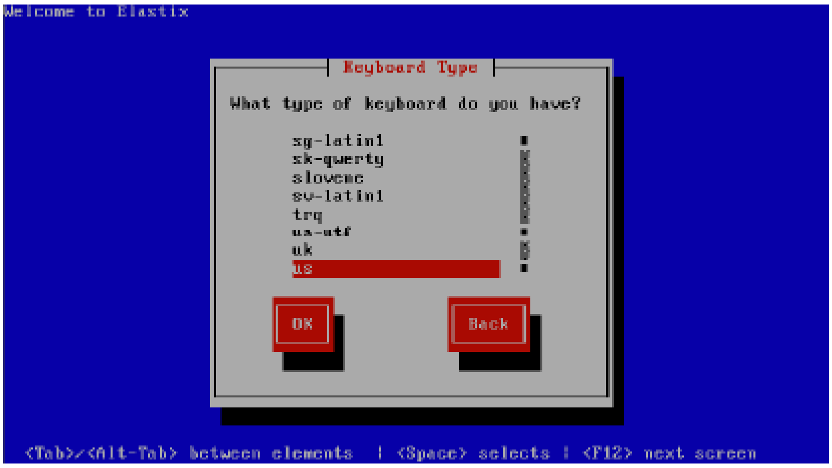
a) The system will go into the CD guide after the PC being started. Then the following interface will be shown on the screen. See Figure 1. Press Enter directly to go into the default installation mode.



b) Next, choose the language for installation. Here select 'English' (Figure 2).

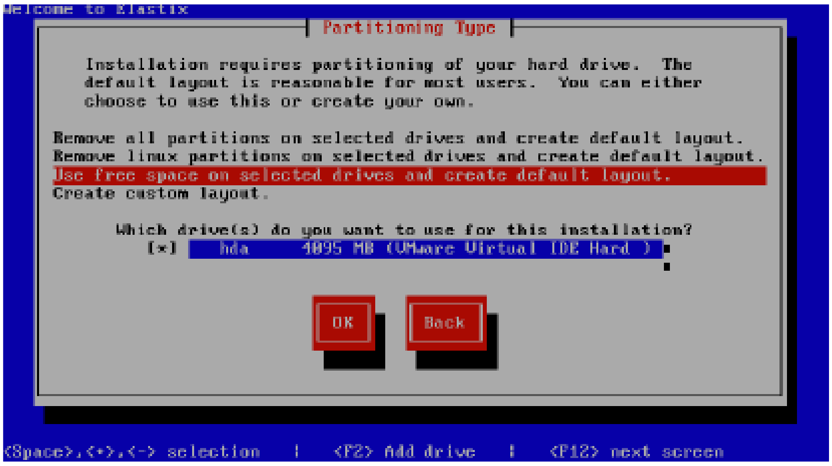


c) Next, choose a keyboard type according to your requirement. Usually we choose 'us' Figure 3).

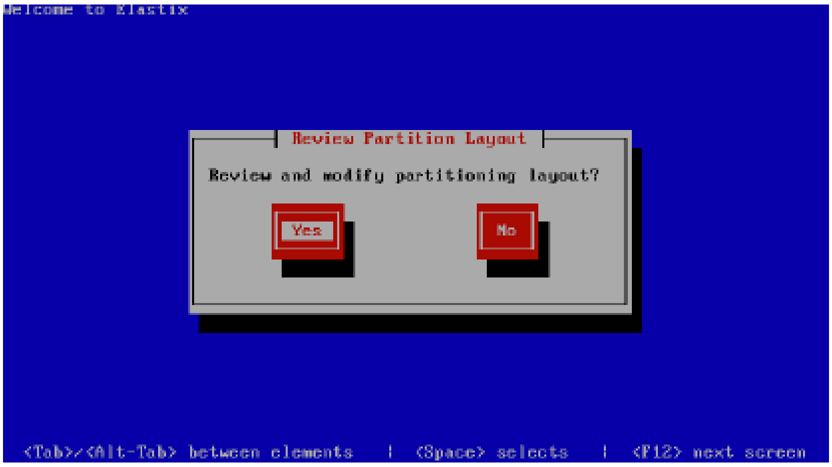


d) Then the welcome interface as shown below appears. Just click on 'OK' (Figure 4).

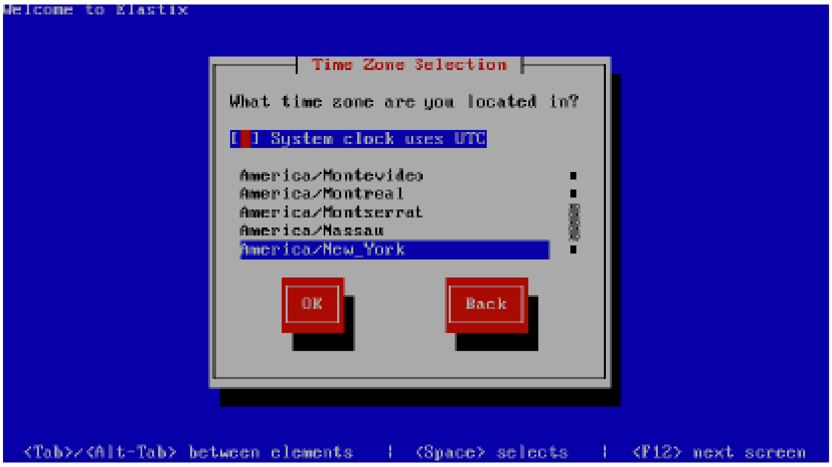
e) Next is the partitioning operation. You have four options to select. For a brand new HD, select the default setting 'Use free space on selected drivers and create default layout'. For an HD with some data already, if you want to discard it, use the option 'Remove all partitions on selected drivers and create default layout'; if you want to keep the old data, select the option ‘Create custom layout’ to do partitioning. What we use here is a new HD. Select the default setting and click on ‘OK’ (Figure 5).



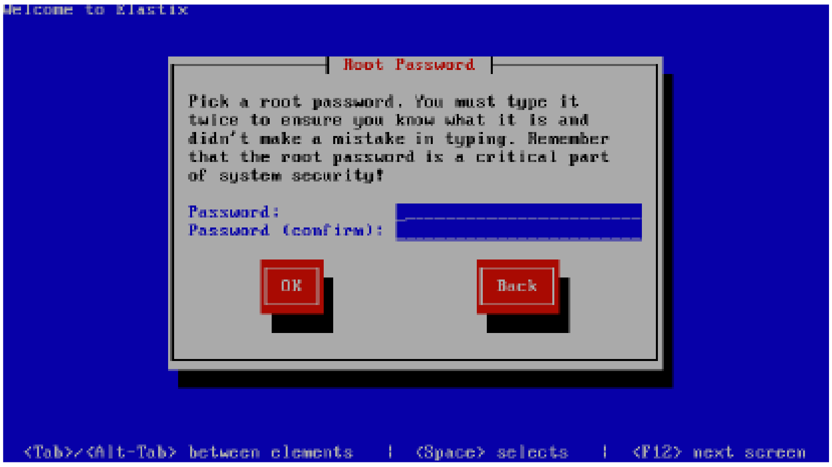
f) Next, the following prompt ‘Review and modify partitioning layout?’ pops up. Select ‘No’ here (Figure 6).



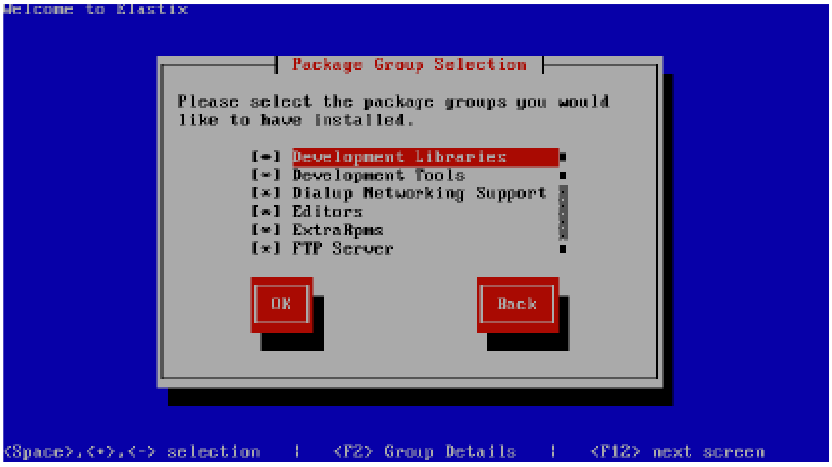
g) Next, select a time zone according to the real situation. Here we select America/New_York’ (Figure 7).



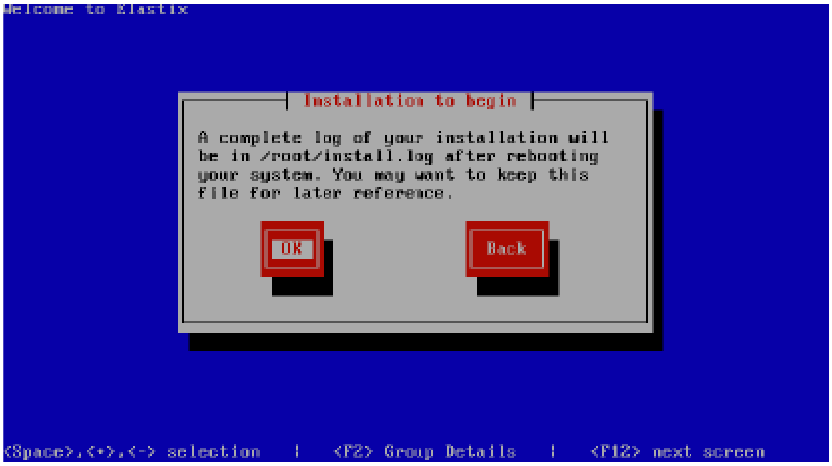
h) Next, enter the system administrator password (Figure 8).



i) Next, choose the packages you need to install. Here we directly select 'OK' (Figure 9).

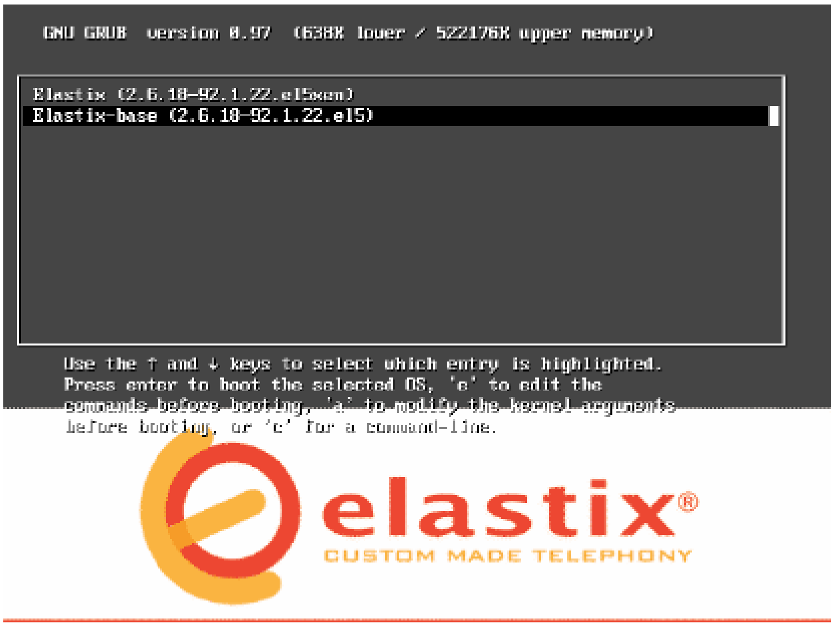


j) Next, a prompt appears to say the installation begins. And a log is accordingly generated to this prompt. Select 'OK' here (Figure 10).



k) Next, the partitioning and formatting of the HD begins. After that, the system installation starts. Upon all files being installed successfully, the PC will be restarted automatically.

l) After the PC restart, the system goes into the startup interface. By default, the system will boot up in Elastix-base mode (Figure 11).



Step3: Log on the system

There pops up the login prompt after the system startup. Please use the root username to log in, and the password is just the one set during the installation process.

Step4: Configure the network

After entering the Elastix system, you should first configure the network so as to connect the system to Internet.

To be exact, run the command 'system-config-network' to configure.

 # system-config-network

On the displayed menu:

Select your network card and press 'Enter'. Then configure the basic information of the network card, like the IP and gateway addresses. After that, exit the interface.

Next, to configure the DNS server, first open the configuration file:

 # vi /etc/resolv.conf

Then add the following line to this configuration file:

nameserver 202.101.172.35

202.101.172.35 is the assumptive address of the DNS server. Please fill in according to your real situation. If you have more than one DNS servers, add multiple lines here.

Save the above configurations and then restart the network service to put them into effective.

 # service network restart

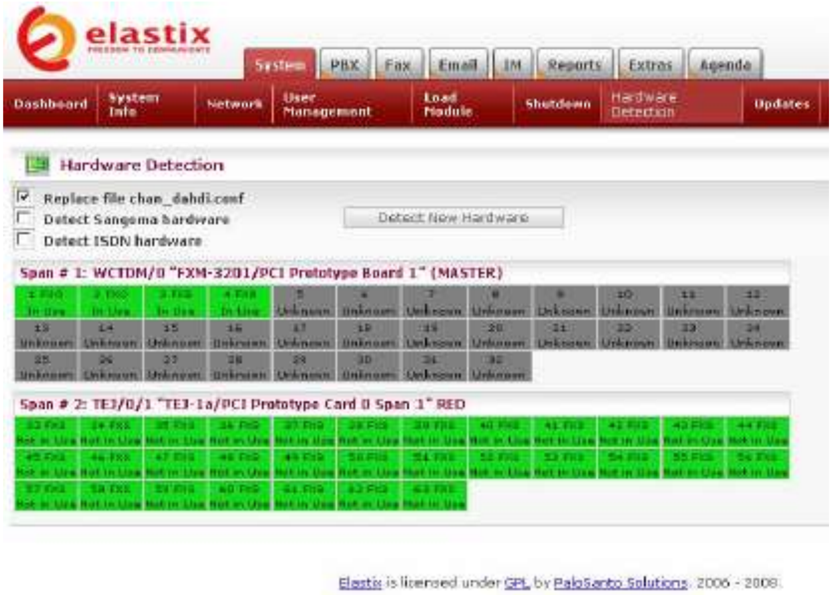
Note: The IP address must be set by configuration tools under the character interface of the local PC. The DNS server, however, can also be set in the WEB mode. To be exact, use another PC to log into this PC in the WEB mode. Use the administrator username 'admin' and the password 'palosanto' to visit the system. Then go to the item 'Network' in the page 'System' to configure the information of the DNS server.

Step5: Configuration and Management of Elastix

In the address bar of the browser, enter the IP address of the Elastix system to go into the initial interface of Elastix Enter the administrator username admin and the password palosanto to reach the configuration and management interface.

On the upward side of the main interface of Elastix is the menu bar

First of all, click on 'Hardware Detection' in SYSTEM menu to detect the installed hardware. In the displayed page (Figure 14), tick the option 'Replace file chan dahdi.conf' and then click the button 'Detect New Hardware' to configure the hardware channels.



Then you can configure the PBX according to Elastix explanation.

Here we use an actual example to explain how to configure.

Take the FXM3201P board as an example. Install an FXM3201P motherboard with an FXO module and an FXS module. Channel 1 and Channel 2 on the board are FXO (trunk) while Channel 3 and Channel 4 are FXS (station). You can see from the above figure that the corresponding trunks in the Elastix system are zap channel 1 and channel 2, the corresponding stations are zap channel 3 and channel 4. If there are multiple boards in the system, the channels are arranged by board number.

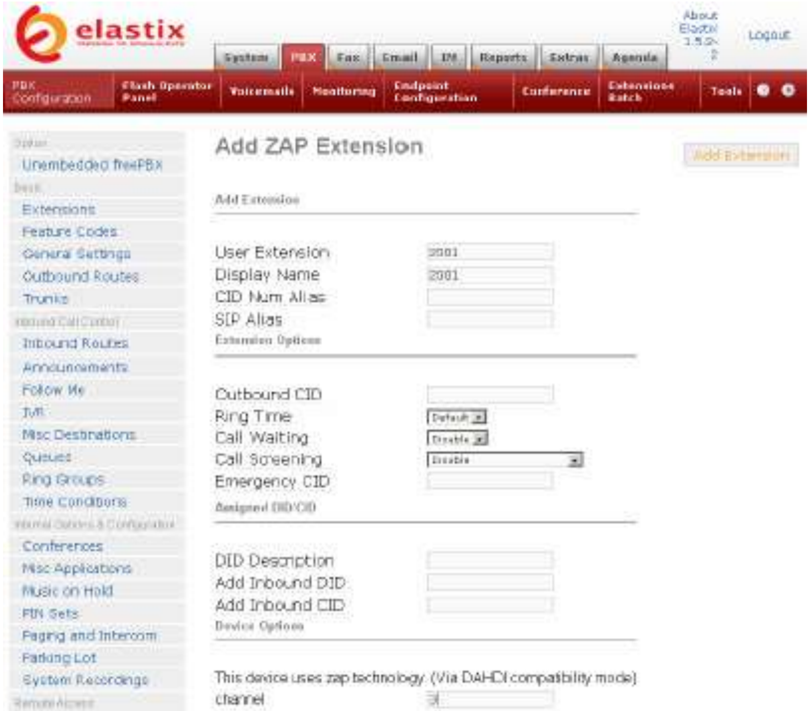
Now we demonstrate such functions as making a call from extension to extension, a call from extension to trunk, and a call from trunk to extension.

First, click on 'PBX' in the menu bar to go by default into the Extensions setting, or click on the menu 'PBX Configuration' and then click the item 'Extensions' in the left navigation bar (Figure 15).

Step1: Configure extensions

In this situation, there are two station channels on the FXM board respectively corresponding to zap channel 3 and zap channel 4. We need to add two Extensions whose numbers are supposed to be 2001 and 2002.

To add the information about the first extension 2001, choose Generic ZAP Device in the pull-down box for Device and press the Submit button to submit (see Figure 15). Then fill in some relative information on the page shown afterwards. Fill in 2001 for both options 'User Extension' and 'Display Name'. Find the sentence 'This device use technology. (Via DAHDI compatibility mode)' (see Figure 16) and fill in 3 for the following option 'channel'. This indicates Extension 2001 uses zap channel 3. Press Submit and the configuration of Extension 2001 is finished.



After that, return to the top 'Add an Extension' to add Extension 2002. Configure it to use zap channel 4. Then both extensions are well configured.

When the modified configuration is submitted, there appears a prompt 'Apply Configuration Changes Here' in red on the top right corner of this page (see Figure 17). Click it to apply the modified configuration. Now we can make calls from extension to extension. Dial 2002 on the first extension to call the second extension.

Step2: Configure trunks

Now there are two trunk channels on the FXM board respectively corresponding to zap channel 1 and zap channel 2.

Click the item Trunks in the left navigation bar. You can see from the right side of this page (see Figure 18) that the default setting has included a trunk. Click 'Trunk ZAP/g0' and you will see the default value of 'Zap Identifier (trunk name)' is g0. Modify it to 1 which indicates this trunk uses zap channel 1 and leave other parameters unchanged. Save the change and the configuration of the first trunk is finished (see Figure 19).

Then add the second trunk. Click 'Add a Trunk' on the right and press 'Add Zap Trunk (DAHDI compatibility mode)' (see Figure 18). Fill in 2 for 'Zap Identifier (trunk name)' which indicates this trunk uses zap channel 2. Click the Submit button to submit. Now both trunks are properly configured (see Figure 19).

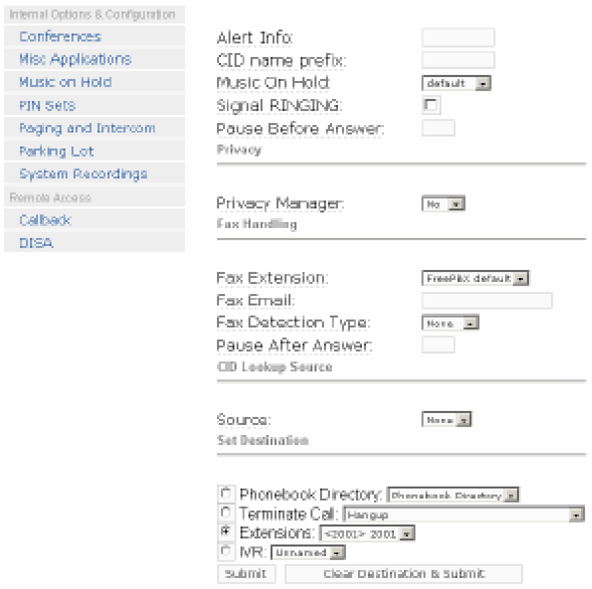
Next, we shall manage to perform the call from extension to trunk.

Step3: Configure the outbound route for calls from extension to trunk

Find the option Basic in the left navigation bar and click Outbound Routes. You can see from the right side of this page that the default setting has included a route with the name of 0 9_outside which indicates the rule to dial 9 before the phone number (see Figure 20). The outbound call is routed on ZAP/1. Actually, dial 9+phone number on the extension and the call will be routed out through zap channel 1. You can modify the configuration and apply it to make calls from extension to trunks.

Step3: Make calls from trunk to extension

To make calls from trunk to extension, you need to configure ‘Inbound Routes’. Find the option Inbound Call Control and select Inbound Routes. Go to the bottom of the displayed page to find Set Destination (see Figure 22). Select Extensions and designate some extension. Thus, when a call comes in from a trunk, the specified extension rings directly. Just pick up the call and talk. Also you can set other inbound routes like IVR to complete corresponding flows. However, the IVR must be set beforehand.



At last don't forget to click 'Apply Configuration Changes' to make modified configurations effective; otherwise, no modification works. Then you can perform call tests based on the above configurations.

Now you are allowed to use the Synway FXM3201P board in the Elastix system to make simple calls.

Points to Remember

- ✓ **Common call routing methods include** Round Robin Call Routing**,** Regular Call Routing**,** Uniform Call Routing**,** Simultaneous Call Routing and Weighted Call Routing.
- ✓ **Elastix** is an open-source unified communications server software that brings together: IP PBX, Email, IM and Faxing. The Elastix functionality is based on open-source projects including Asterisk, HylaFAX, Openfire and Postfix.

Application of learning 3.4:

AGATAKO international secondary school located at Bugesera district want to use VOIP System Sever in their network, they need to hire VoIP Server expert who is responsible to make configuration of their VOIP system server based (IP PBX/PABX/PBX), so you are the one who hired to perform this task.

Indicative content 3.5: Testing of VoIP System (Speed, Latency, Jitter, Packet Loss and QoS)

Duration: 5 hrs

Theoretical Activity 3.5.1: Description of testing VoIP system (VoIP router based and VoIP Server based)

Tasks:

Task 1: In small groups, you are requested to answer the following questions related to the Testing of VoIP system (VoIP router based and VoIP Server based

i. What is VoIP system testing?

ii. In VoIP System testing is based on two criteria, router based, and server based so differentiate those criteria in detail.

Task 2: Provide the answer for the asked questions and write them on papers and present the findings/answers to trainer and whole class.

Task 3: In addition, ask questions where necessary.

Task 4: For more clarification, read the **key readings 3.5.1.**

Key readings 3.5.1.: Description of testing VoIP system (VoIP router based and VoIP Server based)

Testing a Voice over Internet Protocol (VoIP) system, whether it's VoIP router-based for local calls or VoIP server-based for local calls, outgoing/external calls, and call routing, is crucial to ensure reliable and high-quality voice communication.

- **VoIP Router-Based Testing**

✓ **Local Calls**

a. **Network Connectivity**:

- Verify that all VoIP routers are properly connected to the network.
- Ensure that the network infrastructure (routers, switches, etc.) supports the required bandwidth and Quality of Service (QoS) settings for VoIP traffic.

b. **VoIP Router Configuration**:

- Confirm that VoIP routers are configured correctly, including codec settings and network parameters.
- Test call routing within the local network by placing calls between different extensions or devices connected to the same router.

c. **Call Quality Testing**:

- Conduct voice quality tests by making local calls and assessing audio clarity and latency.
- Use tools like PESQ (Perceptual Evaluation of Speech Quality) or MOS (Mean Opinion Score) measurements to quantify voice quality.

d. **Echo and Jitter Testing**:

- Evaluate echo and jitter levels by simulating network conditions that may cause these issues.
- Adjust settings to mitigate echo and jitter, if necessary.

- **VoIP Server-Based Testing**

✓ **Local Calls Testing:**

- Place calls between users on the same VoIP server to test local call functionality.
- Verify that call setup, audio quality, and call termination work as expected.

✓ **Outgoing/External Calls:**

- Configure the VoIP server to connect to external VoIP providers or the public switched telephone network (PSTN).
- Test outgoing calls to external numbers to check for call quality, routing, and billing accuracy.

✓ **Call Routing Testing:**

- Set up and test different call routing scenarios based on your business requirements.

Verify that calls are correctly routed to specific destinations, such as departments, extensions, or IVR systems.

Points to Remember

In the process of VOIP system testing there are key criteria you focus that are "Speed, Latency, Jitter, Packet Loss and QoS" based on **VOIP router** (Focus on local calls) and **VOIP server** (focus on Local calls Outgoing/External calls Call routing).

Application of learning 3.5:

After general assembly of ABANYAMURAVA Cooperative members they have decided to use VOIP System in their communication. They hired a technician from the GU Network Company for installing and configuring that system. So, one issue face, this technician hasn't ability to test her/him configuration, as VOIP System tester you are hired to test that system.

Task1: Test this VOIP System with basing on router configuration.

Task 2: Test this VOIP System with basing on server configuration.

Learning outcome 3 end assessment

Theoretical assessment

Q1. Choose the correct answer.

1. Which protocol is commonly used for signalling in a VoIP system?

A. TCP
B. UDP
C. SIP
D. HTTP

2. What is the primary function of QoS in a VoIP system?

A. Encrypting voice packets
B. Prioritizing voice traffic over other data
C. Compressing voice data
D. Converting analogy to digital signals

3. Which of the following is NOT a common method for assigning phone numbers in a VoIP system?

A. Sequential
B. Random
C. Departmental
D. Extension-based

4. What is the primary purpose of a numbering plan?

A. To assign unique numbers to users
B. To route calls efficiently
C. To manage call queues
D. To provide caller ID information

5. What is the primary function of a Call Manager?

A. Converting analogy signals to digital
B. Routing calls between users
C. Providing power to IP phones
D. Compressing voice data

6. Which of the following is NOT a common feature of a Call Manager?

A. Call forwarding
B. Voicemail
C. Call waiting
D. Network address translation (NAT)

6. Which of the following is NOT a typical function of an IP PBX?

A. Call routing
B. Voicemail
C. Internet browsing
D. Call conferencing

8. What is the primary goal of call routing?
    A. Determining the cost of a call
    B. Selecting the best codec for a call
    C. Directing calls to the correct destination
    D. Preventing call fraud
9. Which of the following is NOT a common call routing method?

A. Direct inward dialling (DID)
B. Hunt groups
C. Voicemail
D. IVR

Q2. Answer by True or False

1. QoS can improve call quality by reducing jitter and packet loss.
2. A VoIP system can operate independently of the internet.
3. An IP PBX can handle both analogy and digital phones.
4. Call routing is essential for large-scale VoIP deployments.
5. A PBX is a type of VoIP system server.
6. CME is a software application used for managing VoIP calls.

Practical assessment

IBYIWACU Manufacturing Company needs to deploy VOIP system in their 30 offices as NIT technician you're hired to make this system. Your responsibility is to select materials, equipment's and tools required to complete this system, installation all equipment in the right position and to make configuration of those selected equipment's. After that you have to make testing and work done report.

END

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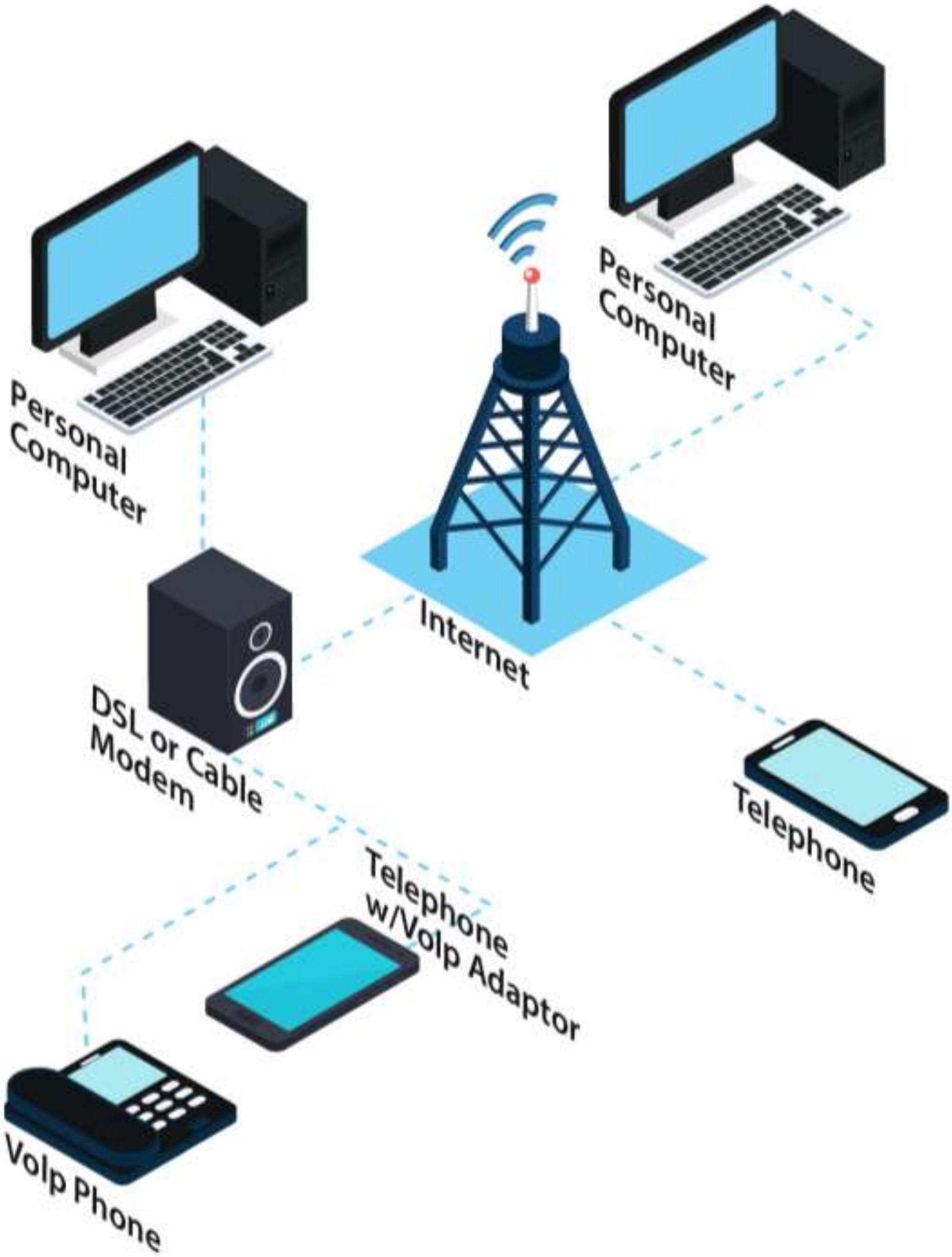
Personal Computer
Personal Computer
Internet
DSL or Cable Modem
Telephone
Telephone w/Volp Adaptor
Volp Phone

Indicative contents
4.1. Description of VoIP system information **4.2. Monitoring of VoIP system** **4.3. Interpretation of VoIP call logs** **4.4. Reporting of VoIP incidents**

Key Competencies for Learning Outcome 4: Operate VOIP System Equipment

Knowledge	Skills	Attitudes
• Description of VoIP system information • Description of monitoring metrics • Description of monitoring methodology • Description of VoIP system monitoring tools/software • Description of VoIP Call logs metrics. • Identification of VoIP Call analytics tools/software • Description of VoIP incident categories • Description of VoIP Incidents reporting tools/software	• Using of VoIP system monitoring tools/software • Using VoIP call analytics tools/software • Using VoIP incidents reporting tools/software.	• Being attentive • Having Curiosity • Heaving teamwork • Being Punctuality • Being Creativity • Being Multitasking • Being Flexibility

Duration: 10 hrs

Learning outcome 4 objectives:

By the end of the learning outcome, the trainees will be able to:

1. Describe properly VoIP system information based on recorded information.
2. Describe properly VoIP monitoring metric according to the system functionality.
3. Use correctly VoIP system monitoring tools/software according to the system functionality.
4. Describe properly monitoring methodology according to the system functionality.
5. Use correctly VoIP call analytics according to the system functionality.
6. Describe properly VoIP system monitoring tools/software according to the system functionality.
7. Describe properly VoIP call logs metrics based on call History.
8. Identify properly VoIP call analytics tools/software based on call History.
9. Describe properly VoIP incidents categories based on call History.
10. Describe properly VoIP incidents reporting tools/software based on type of testing.
11. Use correctly VoIP incidents reporting tools/software based on type of testing.

Resources

Equipment	**Tools**	**Materials**
• Channel bank • IP phone • Power backup (UPS) • IP PBX Server/PBX/PABX Server • VoIP Router • VoIP Gateway • VoIP Headset • PoE • Switch • Computer	• Simulators (Packet tracer, GNS3) • Operating system (Linux)	• Internet bundles

Indicative content 4.1: Description of VoIP System Information

Duration: 3 hrs

Theoretical Activity 4.1.1: Description of VoIP system information.

Tasks:

Task 1: Answer the following questions related to the Describe information type in VoIP system.

i. What are the information types do you know that can be shared through VoIP system?
ii. What are the backup types used for VoIP information system?
iii. Which best backup technology that can be used in VoIP system?

Task 2: Provide the answer for the asked questions and write them on papers and present the findings/answers to trainer and whole class

Task 3: In addition, ask questions where necessary.

Task 4: For more clarification, read the **key readings 4.1.1**

Key readings 4.1.1: Description of VoIP system information

- **Description of VoIP System Information**

✓ **Information types**

- Voice

voice can refer to the transmission of speech over networks, as in voice over IP (VoIP) or voice recognition technology.

- Video

Video is a sequence of visual images that are displayed in rapid succession, creating the illusion of motion.

- Picture

A picture is a static visual representation of an object, scene, or concept. Pictures can be stored and displayed in various digital formats, such as JPEG, PNG, or GIF.

- Text

Text refers to written or printed words that convey information, ideas, or messages. It is a form of communication that has been used for thousands of years and remains an essential means of communication today.

- Email

Email (short for electronic mail) is a digital means of communication that allows users to send and receive messages and files over the internet. It is a fast, efficient, and cost-effective way to communicate with individuals or groups anywhere in the world.

✓ **Description of backup types**

There are mainly three types of backup: full, differential, and incremental.

A full backup is the most complete type of backup where you clone all the selected data. This includes files, folders, SaaS applications, hard drives and more. The highlight of a full backup is the minimal time it requires to restore data. However, since as everything is backed up in one go, it takes longer to backup compared to other types of backup.

A differential backup straddles the line between a full and an incremental backup. This type of backup involves backing up data that was created or changed since the last full backup. To put it simply, a full backup is done initially, and then subsequent backups are run to include all the changes made to the files and folders.

The first backup in **an incremental backup** is a full backup. The succeeding backups will only store changes that were made to the previous backup. Businesses have more flexibility in spinning these types of backups as often as they want, with only the most recent changes stored. Incremental backup requires space to store only the changes (increments), which allows for lightning-fast backups.

✓ **Use of Data backup Techniques/Technologies**

- **Removable media**

Removable media refers to portable storage devices that can be easily detached from a computer or other electronic device. These devices allow users to store and transport data, such as documents, images, videos, or music, from one device to another.

Examples of removable media include USB flash drives, external hard drives, SD cards, CDs, DVDs, and floppy disks. These devices are typically small, lightweight, and durable, making them convenient for use in various settings, such as in the office, at home, or on the go.

- **Redundancy**

Redundancy refers to the duplication or replication of critical components, systems, or processes, with the aim of improving reliability, availability, and fault tolerance.

- **External hard drive**

An external hard drive is a portable storage device that connects to a computer or other electronic device via a cable or wireless connection. It allows users to store, backup, and transport data, such as documents, photos, videos, or music, from one device to another.

- **Backup software**

Backup software is a type of software application that is designed to create copies of data and files, and store them in a secure location, such as an external hard drive or cloud storage, to protect against data loss in case of system failure, cyber-attacks, or other types of disasters.

- **Cloud backup**

Cloud backup, also known as online backup or remote backup, is a service that allows you to store your data remotely on a server that is owned and managed by a third-party provider. This type of backup enables you to protect your important data against loss, theft, or damage by storing it in a secure, off-site location.

Points to Remember

- ✓ **Voice** can refer to the transmission of speech over networks, as in voice over IP (VoIP) or voice recognition technology.
- ✓ **Video** is a sequence of visual images that are displayed in rapid succession, creating the illusion of motion.
- ✓ **A picture** is a static visual representation of an object, scene, or concept. Pictures can be stored and displayed in various digital formats, such as JPEG, PNG, or GIF.
- ✓ **Text** refers to written or printed words that convey information, ideas, or messages. It is a form of communication that has been used for thousands of years and remains an essential means of communication today.
- ✓ **Email** (short for electronic mail) is a digital means of communication that allows users to send and receive messages and files over the internet.

Application of learning 4.1.

You are a network engineer tasked with documenting the existing VoIP system infrastructure for a medium-sized enterprise. The goal is to create a comprehensive inventory of system components, their configurations, and interconnections. This information will be used for troubleshooting, system upgrades, and future planning.

Task 1: Gather detailed information about the VoIP system components, including IP PBX, VoIP phones, gateways, and network equipment.

Task 2: Document the network topology and configuration settings related to VoIP traffic.

Task 3: Create a visual representation of the VoIP system architecture.

Task 4: Identify potential vulnerabilities and areas for improvement.

Task 5: Develop a baseline for future system performance monitoring.

Indicative content 4.2: Monitoring of VoIP System Information

Duration: 3 hrs

Theoretical Activity 4.2.1: Description of monitoring metrics

Tasks:

Task 1: In small groups, you are requested to answer the following questions related to the Description of monitoring metrics

i. What are the monitoring metrics do you know?
ii. What are the backup monitoring methodologies used in VoIP system?
iii. What is the Use of VoIP System Monitoring tools/software?

Task 2: Provide the answer for the asked questions and write them on papers and then present the findings/answers to the whole class

Task 3: In addition, ask questions where necessary.

Task 4: For more clarification, read the **key readings 4.2.1.**

Key readings 4.2.1: Description of monitoring metrics

- **Jitter**

Jitter is a term used to describe the variation or inconsistency in the timing of a signal. It is most used to describe digital signals and is often caused by issues with the transmission or processing of the signal.

- **Latency**

Latency refers to the delay or time lag between the transmission of a signal and its receipt or processing by the recipient. It is commonly used in the context of digital communications, such as networks, servers, and internet connections.

- **Packet loss**

Packet loss is a term used to describe the failure of one or more packets of data to reach their intended destination in a network communication.

- **Quality of services for Voice**

Quality of Service (QoS) is a set of networking protocols and technologies that aim to provide reliable and predictable performance for voice and other real-time applications over a network.

✓ **Description of monitoring methodology**

- **Data collection**

Data collection is the process of gathering and measuring information on a specific topic or subject, using various methods and tools.

The goal of data collection is to obtain accurate and relevant data that can be analysed and used to make informed decisions or to gain insights into a particular issue or phenomenon.

- **Data aggregation**

Data aggregation is the process of combining and summarizing multiple pieces of data into a single cohesive unit. The goal of data aggregation is to simplify complex data sets and make them more easily understandable and useful for analysis and decision-making.

- **Data analysis**

Data analysis is the process of systematically examining and interpreting data to extract meaningful insights and draw conclusions. It involves a range of techniques and methods for identifying patterns, trends, relationships, and other useful information within data sets.

✓ Use of VoIP System Monitoring tools/software

VoIP monitoring is the tracking of different metrics including jitter, latency, and packet loss, all of which affect the quality of a VoIP call. Simply put, it is monitoring the quality of service (QoS) of VoIP calls, which combines both fault and performance management. Tracking metrics from the source to the destination and vice versa along with mean opinion score (MOS) and round-trip time (RTT) will help to ensure that everything is under control throughout communication and in the connection. To measure VoIP Quality for VoIP Monitoring, specialized network monitoring tools can capture and analyse VoIP traffic to measure the key metrics that impact call quality. These metrics can be monitored in real-time, and businesses can set thresholds for acceptable values for each metric.

VoIP Monitoring tools/software's

1.Dotcom-Monitor is a VoIP Monitoring service that services allow you to instantly troubleshoot phone number availability from anywhere in the world, at any time!

2. Pressler PRTG Network Monitoring: It helps monitor traffic, serves as a troubleshooter, reduces VoIP errors, minimizes latency, packet loss, or jitter, and improves its performance.

3. Site24x7 It is easier to troubleshoot the quality issues in real-time with performance metrics and available stats giving deep insights.

4. SolarWinds: It helps identify any distortion, noise, and latency while eliminating problems. The tool can deliver control and central visibility to the users and get historical and real-time statistics reporting.

5. VoIP Spear: VoIP Spear tracks the voice quality to ensure that one can analyze it and fix the issues. The software helps monitor the quality of the VoIP 24×7 to see the status of the problems. VoIP Spear offers a score showcasing the quality of calls as the Mean Opinion Score (MOS) using the chart.

6. Riverbed UCExpert is an easy-to-use VoIP monitoring tool where users can easily interact and collaborate with customers and teams. It is a tool that improves productivity by reducing help desk calls and effective collaboration. One can enhance voice and video collaboration applications, messaging capabilities, social media, and mobile apps.

7. ExtraHop can instantly monitor all the data to understand the IT activities that might affect its quality. It also isolates the user issues and helps offer an understanding of conditions. The software supports compliance and auditing, correlates traffic, analyzes traffic, and gains visibility.

8. Manageengine OpManager is a top-notch tool that ensures zero complaints on the call quality, brings the network together, service monitoring, performance and fault performance management, and network traffic analysis. The tool’s features include proactively monitoring VoIP performance and VoIP quality across WAN infrastructure.

Points to Remember

- ✓ **Jitter** is a term used to describe the variation or inconsistency in the timing of a signal. It is most used to describe digital signals and is often caused by issues with the transmission or processing of the signal.
- ✓ **Latency** refers to the delay or time lag between the transmission of a signal and its receipt or processing by the recipient.
- ✓ **Packet loss** is a term used to describe the failure of one or more packets of data to reach their intended destination in a network communication.
- ✓ **Quality of Service (QoS)** is a set of networking protocols and technologies that aim to provide reliable and predictable performance for voice and other real-time applications over a network.

Application of learning 4.2.

Your organization relies heavily on its VoIP system for daily operations. To ensure business continuity and minimize disruptions due to system failures or disasters, you are tasked with evaluating and implementing appropriate backup strategies.

Task 1: Identify different types of backups applicable to VoIP systems.
Task 2: Assess the advantages and disadvantages of each backup type.
Task 3: Determine the optimal backup strategy based on the organization's needs and resources.
Task 4: Develop a backup schedule and retention policy.
Task 5: Implement backup procedures and test their effectiveness.

Indicative content 4.3: Interpretation of VoIP Call Logs

Duration: 2 hrs

Theoretical Activity 4.3.1: Description of VoIP Call logs metrics

Tasks:

Task 1: Answer the following questions related to the Description of VoIP Call logs metrics

i. What are the VoIP call log metrics do you know?

ii. What are the Analytic tools used in VoIP system?

iii. What is the Use of VoIP System Monitoring tools/software?

Task 2: Provide the answer for the asked questions and write them on papers and then present the findings/answers to the whole class.

Task 3: In addition, ask questions where necessary

Task 4: For more clarification, read the **key readings 4.3.1.**

Key readings 4.3.1: Description of VoIP Call logs metrics

- **Date and time of a telephone call**

The date and time of a telephone call refer to the exact time and date when a phone call was made or received. This information can be useful for several purposes, including tracking call activity, billing, and legal or investigative purposes.

- **Call duration (in minutes)**

Call duration refers to the length of time that a phone call lasts, typically measured in minutes or seconds.

The call duration starts from the moment the call is connected and ends when the call is disconnected by either party.

- Information on the source and destination telephone numbers
- Type of a phone call – toll-free, inbound, outbound Cost of a call (per minute rate)

- **Picked Calls**

"Picked calls" refers to the number of incoming phone calls that are answered by the recipient or the intended person. In other words, when someone receives a phone call and answers it, the call is considered a picked call.

- **Holded calls**

"Holded calls" or "held calls" refer to incoming phone calls that are placed on hold by the recipient or the intended person. When someone receives a phone call but is not able to immediately answer it, they may put the call on hold to attend to other matters or to transfer the call to another person or department.

- **Routed calls**

A routed call refers to a phone call that is directed or "routed" from one point to another over a network of interconnected communication devices.

- **Recorded calls**

Recorded calls refer to phone conversations or other types of audio communication that have been digitally recorded and stored for later playback or review.

✓ Use of VoIP Call analytics tools/software

VoIP call analytics is a built-in data reporting system that comes along with many VoIP phone systems today. It is extremely beneficial for businesses who manage many outbound/inbound customer calls on a daily basis. Call analytics can be pulled easily from your VoIP dashboard on your desktop or mobile application.

VoIP analysis tools allow you to view a summary of overall VoIP status, drill down to connection details for effective troubleshooting, identify quality issues within the network, and measure call quality.

Some key features of common VoIP analysis tools include:

- Real-time VoIP monitoring, such as call control, call status, call destination, and bandwidth consumption.
- Maintenance of QoS level according to industry standards and management of VoIP audio quality
- Detailed views with in-depth call detail records
- Detection of latency, jitter, and packet loss for calls made between two distributed sites.
- Comparison of VoIP to overall network performance and analysis of bandwidth utilization to determine if the network load is affecting call quality.
- Analysis of quality score impairment factors, such as loudness, talker echo, circuit noise, floor noise, and room noise
- Detailed view of information on conversation streams, including call setup communications, the actual voice conversation, and call teardown communications
- Decoding and reconstructing voice and video streams for replay
- Diagnostics and assistance in troubleshooting connection issues

If you experience call quality issues, an analysis tool can help identify the problematic calls and isolate their paths, so technicians can perform further analyses on the switches, routers, and subsystems to improve performance.

Besides accurately tracking call information and performance, a VoIP analysis tool should provide good usability and navigability. You should be able to easily locate the data you need, see it in a clear and straightforward view, and manipulate it for further interpretation.

In addition, consider the reporting capability of a VoIP analysis tool to make sure it suits your needs. The ability to visualize the data and graphically display VoIP analysis results can help your IT team establish a baseline and troubleshoot more efficiently.

Points to Remember

- ✓ **The date and time of a telephone call** refer to the exact time and date when a phone call was made or received. This information can be useful for several purposes, including tracking call activity, billing, and legal or investigative purposes.
- ✓ **Call duration** refers to the length of time that a phone call lasts, typically measured in minutes or seconds. The call duration starts from the moment the call is connected and ends when the call is disconnected by either party.
- ✓ "**Picked calls"** refers to the number of incoming phone calls that are answered by the recipient or the intended person. In other words, when someone receives a phone call and answers it, the call is considered a picked call.

- ✓ **"Holded calls"** or "held calls" refer to incoming phone calls that are placed on hold by the recipient or the intended person. When someone receives a phone call but is not able to immediately answer it, they may put the call on hold to attend to other matters or to transfer the call to another person or department.
- ✓ **A routed call** refers to a phone call that is directed or "routed" from one point to another over a network of interconnected communication devices.
- ✓ **Recorded calls** refer to phone conversations or other types of audio communication that have been digitally recorded and stored for later playback or review.

Application of learning 4.3.

YYY organization has experienced an increase in customer complaints regarding call quality and service interruptions. To investigate the root cause of these issues, you have been tasked with analysing VoIP call logs.

Task 1: Identify key metrics and data points within VoIP call logs.

Task 2: Analyse call logs to identify patterns and trends related to call quality issues.

Task 3: Correlate call log data with network performance metrics to pinpoint potential problems.
Task 4: Generate reports based on call log analysis to support troubleshooting and performance optimization.

Indicative content 4.4: Reporting of VoIP Incidents

Duration: 2 hrs

Theoretical Activity 4.4.1: Description of VoIP incident categories

Tasks:

Task 1: Answer the following questions related to the Description of VoIP incident categories

i. What are the VoIP incident categories do you know?

ii. What are the incident reporting tools used in VoIP system?

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to trainer and whole class

Task 4: In addition, ask questions where necessary.

Task 5: For more clarification, read the **key readings 4.4.1.**

Key readings 4.4.1.: Description of VoIP incident categories

✓ **Description of VoIP incident categories**

- **Poor call quality:** Bad call quality can be described as choppy audio (your customer might say, “you're breaking up”), a delayed voice transmission (symptoms include talking over each other) or dropped calls. To sum it up, the call quality issues end up affecting the quality of your service to your customers.
- **Sign on issues:** your sign-in may be blocked if the device or location you're using is new. Try again from a device or location that you commonly sign in from.
- **Slow load time:** Being aware of how quickly your website’s pages are loading is crucial. Slow loading times upset existing users, increase bounce rates, affect search engine rankings, and have a serious impact on revenue. So, tracking your site’s loading times, and understanding problems that affect the speed of your website, can be the difference between a failing website and a successful online business.
- **Batch job errors:** A batch job is a predefined group of processing actions submitted to the system to be performed with little or no interaction between the user and the system. Jobs that do not require user interaction to run can be processed as batch jobs.

✓ **Use of VoIP Incidents reporting tools/software**: Incident reporting is the process of notifying a user or administrator of an abnormal event, process or action identified on a computing device, system or environment.

It is part of the security incident and event management (SIEM) process that alerts and logs all security incidents discovered within an IT environment.
Incident reporting is also known as security incident reporting or incident tracking.

VoIP Software with Reporting/Analytics

- ULTATEL Cloud Business Phone System. 4.7. (30) ULTATEL | Phone and Teleommunication Solutions For Business. ...
- GoTo Meeting. Highly viewed. 4.4. (11.4K) ...
- Webex. Highly viewed. 4.4. ...
- VoIP.ms. 4.8. (687) ...
- Zoho Meeting. 4.6. (790) ...
- Talkdesk. 4.5. (718) ...
- RingCentral MVP. 4.2. (1.1K) ...
- GoTo Connect. 4.5. (657)

Points to Remember

- ✓ **Poor call quality**: Bad call quality can be described as choppy audio (your Customer might say, “you're breaking up”), a delayed voice transmission (symptoms include talking over each other), or dropped calls.
- ✓ **Sign on issues**: your sign-in may be blocked if the device or location you're using is new. Try again from a device or location that you commonly sign in from.
- ✓ **Slow load time:** Being aware of how quickly your website’s pages are loading is crucial. Slow loading times upset existing users, increase bounce rates, affect search engine rankings, and have a serious impact on revenue.
- ✓ **Batch job errors:** A batch job is a predefined group of processing actions submitted to the system to be performed with little or no interaction between the user and the system.

Application of learning 4.4.

Your organization has experienced a series of VoIP-related incidents, including service outages, call quality issues, and security breaches. To improve incident response and prevent recurrence, you are tasked with developing a comprehensive incident reporting and management process.

Tasks

1. Establish clear incident reporting procedures for end-users and IT staff.
2. Develop incident classification and prioritization criteria.
3. Implement an incident tracking and management system.
4. Analyse incident data to identify trends and root causes.
5. Develop preventive measures to reduce incident frequency.

Learning outcome 4 end assessment

Theoretical assessment

Q1. Choose the correct answer.

i. Which of the following is NOT a core component of a VoIP system?

A. Analog Telephone Adapter (ATA)
B. IP PBX
C. Codec
D. Router

ii. What is the primary function of a codec in a VoIP system?

A. Converting analogy signals to digital
B. Routing calls between users
C. Providing power to IP phones
D. Compressing and decompressing voice data

iii. Which metric is used to measure the quality of voice transmission in a VoIP system**?**

a. Mean Opinion Score (MOS)
b. Packet Loss Rate
c. Jitter
d. All the above

iv. What is the primary goal of VoIP system monitoring?

A. Identifying potential security threats
B. Ensuring optimal call quality
C. Reducing operating costs
D. Increasing user satisfaction

v. Which of the following is NOT a typical information included in a VoIP call log?

a. Call duration
b. Caller ID
c. Network latency
d. Call termination status

vi. What is the primary purpose of analysing VoIP call logs?

A. Identifying network congestion
B. Improving call quality
C. Preventing fraud
D. All the above

Q2. Answer by True or False

i) VoIP systems can operate over both wired and wireless networks.
ii) Jitter is a measure of the variation in packet arrival times.
iii) Packet loss can significantly impact call quality.
iv) VoIP call logs can be used to identify system performance trends.
v) Regular monitoring of a VoIP system is essential for maintaining service quality.
vi) Incident reports should include detailed information about the problem and its impact.

Practical assessment

You are a network administrator for a small to medium-sized business. The company has recently implemented a VoIP system to replace its traditional analog phone system. You are responsible for the day-to-day operation and maintenance of the VoIP system.

Task:

1. Add, modify, and delete user accounts, assign extensions, and configure phone features.
2. Diagnose and resolve common VoIP problems such as call quality issues, dropped calls, and registration failures.
3. Track key performance indicators (KPIs) like call volume, call duration, and system uptime.
4. Implement security measures to protect the VoIP system from unauthorized access and attacks.
5. Assist employees with VoIP phone usage and troubleshoot issues.

END

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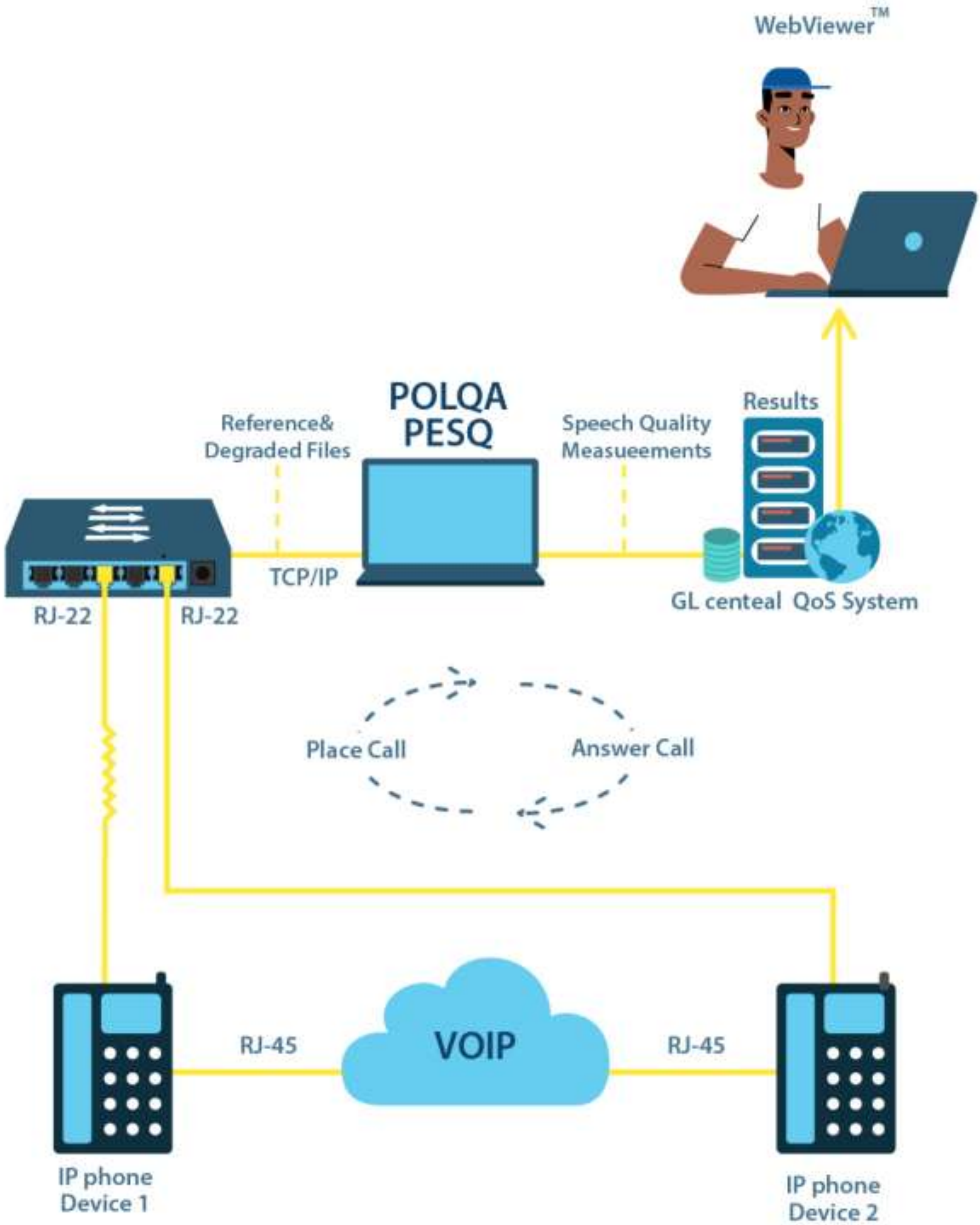
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QoS System
Place Call
Answer Call
RJ-45
VOIP
RJ-45
IP phone
Device 1
IP phone
Device 2

Indicative contents
5.1. Description of VoIP Testing tools **5.2. Description of VoIP testing metrics** **5.3. Testing of VoIP Audio quality** **5.4. Testing of VoIP Video quality** **5.5. Testing of VoIP Routing connectivity** **5.6. Elaboration of VoIP Testing report** **5.7. Description of testing report elements**

Key Competencies for Learning Outcome 5: Test VoIP system

Knowledge	Skills	Attitudes
• Description of VoIP Testing tools • Description of VoIP testing metrics • Description of audio quality metrics • Description of VoIP audio quality testing tools • Description of video quality testing metrics • Description of VoIP video quality testing tools • Description of connectivity testing tools • Description of testing cycle • Description of testing techniques • Description of testing report elements	• Using testing tools. • Selecting testing metrics. • Using VoIP audio quality testing tools • Using of VoIP video quality testing tools • Performing connectivity testing tools •	• Being attentive • Having Curiosity • Having teamwork • Being Punctuality • Being Creativity • Being Multitasking • Being flexibility

Duration: 10 hrs

Learning outcome 5 objectives:

By the end of the learning outcome, the trainees will be able to:

1. Describe properly testing tools based on type of testing.
2. Use appropriately testing tools based on type of testing.
3. Describe properly VoIP testing metrics based on type of testing.
4. Select properly VoIP testing metrics based on type of testing.
5. Describe properly audio quality metric according to the configured VoIP system.
6. Describe properly VoIP audio quality testing tools according to the configured VoIP system.
7. Use appropriately VoIP audio quality testing tools according to the configured VoIP system.
8. Describe properly video quality testing metrics video quality testing metrics.
9. Describe properly VoIP video quality testing tools according to configured VoIP system.
10. Use appropriately video quality testing tools according to configured VoIP system.
11. Describe correctly connectivity testing tools according to the configured VoIP system.
12. Perform connectivity testing tools according to the configured VoIP system.
13. Describe testing cycle based on test results.
14. Describe properly testing techniques based on test results.
15. Describe properly testing report elements based on test results.

Resources

Equipment	Tools	Materials
• Channel bank • IP phone • Power backup (UPS) • IP PBX Server/PBX/PABX Server • VoIP Router • VoIP Gateway • VoIP Headset	• Audio quality testing tools • Video quality testing tools • Connectivity testing tools • Simulators (Packet tracer, GNS3) • Operating system (Linux)	• Internet bundles

Indicative content 5.1: Description of VoIP Testing Tools

Duration: 2 hrs

Theoretical Activity 5.1.1: Description of VoIP Testing tools

Tasks:

Task 1: Answer the following questions related to the Description of VoIP Testing tools

i. What are the VoIP testing tool do you know?
ii. What are the advantages of VoIP testing?
iii. What is the VoIP audio quality?
iv. What is the VoIP video quality?
v. What is routing connectivity?

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to trainer and the whole class

Task 4: In addition, ask questions where necessary.

Task 5: For more clarification, read the **key readings 5.1.1.**

Key readings 5.1.1: Description of VoIP Testing tools

What is a VoIP test?

A VoIP test is a diagnostic tool that measures the quality of your internet connection and evaluates its ability to support VoIP calls. It checks various parameters like bandwidth, latency, jitter, packet loss, and MOS (Mean Opinion Score) to determine the quality of your internet connection.

Furthermore, if your calls are failing altogether, you can troubleshoot by running a VoIP test - an app that measures the speed, jitter and latency of your internet connection. Also known as a VoIP speed test, it's a way to check if your bandwidth is enough to support VoIP calling.

A VoIP test helps you review the overall health of your network connection within a matter of seconds. It also enables you to determine whether the call traffic in your existing phone system is high enough to justify an upgrade.

Examples of VoIP Testing tool

1. **Ping-test.net**: This is a VoIP testing tool which helps in measuring the download and upload speed of the active Internet connection.

2. **Speed test: this is a VoIP tool which has purpose of** measuring your current connection's maximum speed – how fast your device can upload and download information – by accessing nearby test servers.
3. ZDA NET broadband speed test offers a visually pleasing tool that provides details in a neat graphical form that is easy to consume and appealing.

All you need to do is to enter the relevant details into the tool, either from your home or office and provide the postcode. It will then test your VoIP connection and provide you the QoS report.

4. 8×8 VoIP Test passes simulated VoIP traffic to your computer by opening a socket connection to your browser and this, in turn, helps in measuring the performance and quality of your Internet Connection.
5. FreeOLa this tool is able to test the quality of line over variety parameters for instance network latency,download/upload speeds,delay,jitter and packet loss.

Points to Remember

A VoIP test is a diagnostic tool that measures the quality of your internet connection and evaluates its ability to support VoIP calls. It checks various parameters like bandwidth, latency, jitter, packet loss, and MOS (Mean Opinion Score) to determine the quality of your internet connection.

Application of learning 5.1.

ZZZ organization is expanding its VoIP network and requires a comprehensive testing strategy to ensure optimal performance and reliability. To achieve this, you need to identify and understand various VoIP testing tools available in the market.

Tasks:

1. Research and identify different types of VoIP testing tools.
2. Analyse the capabilities and limitations of each tool.
3. Compare and contrast different testing tools based on specific testing requirements.
4. Select appropriate testing tools for different VoIP system components and functionalities.
5. Develop a VoIP testing plan utilizing the chosen tools.

Indicative content 5.2: Description of VoIP Testing Metrics

Duration: 2 hrs

Theoretical Activity 5.2.1: Description of VoIP testing metrics

Tasks:

Task 1: Answer the following questions related to the description of electric quantities

i. What are the VoIP testing metrics do you know?

ii. How can you overcome the VOIP metrics?

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to trainer and the whole class

Task 4: In addition, ask questions where necessary.

Task 5: For more clarification, read the **key readings 5.2.1.**

Key readings 5.2.1: Description of VoIP testing metrics

- ✓ **Jitter** is a term used to describe the variation or inconsistency in the timing of a signal. It is most used to describe digital signals and is often caused by issues with the transmission or processing of the signal.
- ✓ **Latency** refers to the delay or time lag between the transmission of a signal and its receipt or processing by the recipient. It is commonly used in the context of digital communications, such as networks, servers, and internet connections.
- ✓ **Mean opinion score** (MOS) this has been a commonly used metric to measure the overall voice call quality for decades. MOS score is a rating from 1 to 5 of the perceived quality of a voice call, 1 being the lowest score and 5 the highest for excellent quality.
- ✓ **Packet Loss**: Voice and audio data travel in packets across the Internet. Packet loss occurs when one or more packets of data fail to reach their destination. With VoIP calls, packet loss can degrade the quality of calls by dropping parts of the conversation, causing choppy audio and dropped calls.

Points to Remember

- **Jitter** is a term used to describe the variation or inconsistency in the timing of a signal. It is most used to describe digital signals and is often caused by issues with the transmission or processing of the signal.
- **Latency** refers to the delay or time lag between the transmission of a signal and its receipt or processing by the recipient. It is commonly used in the context of digital communications, such as networks, servers, and internet connections.
- **Mean opinion score** (MOS) this has been a commonly used metric to measure the overall voice call quality for decades. MOS score is a rating from 1 to 5 of the perceived quality of a voice call, 1 being the lowest score and 5 the highest for excellent quality.
- **Packet Loss**: Voice and audio data travel in packets across the Internet. Packet loss occurs when one or more packets of data fail to reach their destination.

Application of learning 5.2.

WWWS organization has deployed a VoIP system and is experiencing issues with call quality. To identify the root cause of these problems, you need to establish a comprehensive set of metrics to measure VoIP performance.

Tasks:

1. Identify key performance indicators (KPIs) for VoIP systems.
2. Define relevant metrics to measure call quality, network performance, and system availability.
3. Understand the relationship between VoIP metrics and overall user experience.
4. Develop a methodology for collecting and analysing VoIP metrics.

Indicative content 5.3: Testing of VoIP Audio Quality

Duration: 2 hrs

Theoretical Activity 5.3.1: Description of audio quality metrics

Tasks:

Task 1: Answer the following questions related to the Description of audio quality metrics

i. What are the VoIP audio quality metrics do you know?

ii. What are the VoIP quality testing tools you know to use?

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to trainer and the whole class.

Task 4: In addition, ask questions where necessary.

Task 5: For more clarification, read the **key readings 5.3.1.**

Key readings 5.3.1: Description of audio quality metrics

✓ **Clarity**

Clarity refers to the intelligibility and distinctness of the audio signal.

It encompasses factors such as:

- **Speech intelligibility:** How well can the listener understand the spoken words?
- **Background noise:** The level of unwanted sounds that interfere with the conversation.
- **Echo and reverberation:** The presence of reflected sound that can degrade audio quality.
- **Frequency response:** The range of audible frequencies accurately reproduced.

✓ **Call Delay**

Call delay, also known as latency, is the time elapsed between the transmission of an audio signal and its reception by the listener. It is measured in milliseconds (ms).

Excessive call delay can hinder real-time communication by causing:

- **Echo:** Delayed audio signals can be reflected to the sender, creating an echo effect.
- **Interrupted speech:** Significant delays can disrupt the natural flow of conversation.

- **Difficulty in following conversations:** Multiple participants can lead to overlapping speech and confusion.

✓ **Responsiveness**

Responsiveness is closely related to call delay but focuses on the perceived speed of the audio interaction. It encompasses:

- **Real-time feel:** The sense that the conversation is happening in real-time without noticeable delays.
- **Interactivity:** The ability to engage in a natural and fluid conversation.
- **User satisfaction:** The overall perception of the audio experience.

While call delay is a measurable quantity, responsiveness is often subjective and influenced by factors such as network conditions, codec efficiency, and user expectations.

✓ **Use of VoIP audio quality testing tools**

- **Cable Tester** is an electronic device used to verify the electrical connections in a signal cable or other wired assembly. Basic cable testers are continuity testers that verify the existence of a conductive path between ends of the cable and verify the correct wiring of connectors on the cable.
- **Polarity Checker** In the context of electricity installations, a polarity test is used to confirm the correct connection of the line and neutral conductors.
- Audio Analyzer is a test and measurement instrument used to objectively quantify the audio performance of electronic and electro-acoustical devices.
- Speaker/Headphone Calibration Software Audio Calibration is the process of making micro adjustments in your audio system so that you attain the most pristine sound when you are sitting in the prime location of our surround sound system.
- Bluetooth Interface is a technology standard used to enable short-range wireless communication between electronic devices. Since Bluetooth operates on radio frequencies, rather than the infrared spectrum used by traditional remote controls, devices using this technology do not have to maintain a line of sight to communicate.

Points to Remember

- **Delay** is caused when packets of voice data take more time than expected to reach their destination, Propagation delay is the time it takes for the first bit of a packet to travel from source to destination; jitter is the variation in arrival rate of data packets.
- **Responsiveness** VoIP call quality is the clarity and responsiveness of voice calls made through internet packet delivery.
- **Clarity** VoIP call quality is the clarity and responsiveness of voice calls made through internet packet delivery.

Application of learning 5.3.

YYY organization has recently implemented a new VoIP system and is experiencing issues with call quality. To identify and resolve these problems, you are tasked with conducting a comprehensive audio quality assessment.

Tasks:

1. Define key audio quality metrics relevant to VoIP.
2. Develop test plans to measure audio quality under various network conditions.
3. Utilize appropriate testing tools and methodologies.
4. Analyse test results to identify areas for improvement.
5. Implement corrective actions to enhance audio quality.

Indicative content 5.4: Testing of VoIP Video Quality

Duration: 1 hrs

Theoretical Activity 5.4.1: Description of video quality testing metrics

Tasks:

Task 1: Answer the following questions related to the Description of video quality testing metrics

i. What are the VoIP video quality metrics do you know?

ii. What are the VoIP video quality testing tools you know to use?

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to trainer and the whole class

Task 4: In addition, ask questions where necessary.

Task 5: For more clarification, read the **key readings 5.4.1.**

Key readings 5.4.1: Description of video quality testing metrics

Video Quality Metric (VQM) is a measurement of video quality that combines objective factors (such as resolution, bit rate, and frame rate) with subjective factors (such as perceived image quality) to create a single, unitless score.

This can be helpful in comparing different video encoding settings or codecs. Video quality can be evaluated objectively (by mathematical models) or subjectively (by asking users for their rating). Also, the quality of a system can be determined offline (i.e., in a laboratory setting for developing new codecs or services), or in-service (to monitor and ensure a certain level of quality).

- ✓ **Frame rate**

The frame rate is evaluated using the marker-based method. Each frame in the test video is overlaid with a unique marker that stores the value of the frame sequence number. The frame rate tool decodes each marker in the recorded video and calculates frame rate at a specified period.

If the view-through rate is higher, that means the entire video is engaging; if the rate is lower, it means the video is not successful in engaging audiences. Watch time: Like the view-through rate, the watch time metric is the total amount of time a video has been watched.

✓ **Quality of image**

Image quality is assessed using BRISQUE (Blind/Reference less Image Spatial Quality Evaluator) algorithm. BRISQUE is a no-reference image quality evaluation algorithm. Image resolution is typically described in PPI, which refers to how many pixels are displayed per inch of an image. Higher resolutions mean that there more pixels per inch (PPI), resulting in more pixel information and creating a high-quality, crisp image. Image Metrics is a leading augmented reality company at the forefront of real time human detection and analysis research.

- **Use of VoIP video quality testing tools**

Video streaming test feature is designed to measure video quality of experience on smart mobile device connected over 2G/3G/4G/5G/Wi-Fi networks by calculating key performance indicators for video QoE parameters such as Launch Time, Load Time, Stalled Time and Total Video Play Time. Our solution is highly cost effective (check our prices) and no need to purchase additional hardware when compared to other video streaming performance testing tools.

Points to Remember

- ✓ **VQM** is a measurement of video quality that combines objective factors (such as reso lution, bit rate, and frame rate) with subjective factors (such as perceived image qual ity) to create a single, unitless score.
- ✓ The **frame rate** is evaluated using the marker-based method. Each frame in the test v ideo is overlaid with a unique marker that stores the value of the frame sequence nu mber.
- ✓ **Quality of image**: Image quality is assessed using BRISQUE (Blind/Referenceless Imag e Spatial Quality Evaluator) algorithm. BRISQUE is a no-reference image quality evalu ation algorithm.

Application of learning 5.4.

BBB organization has recently implemented a video conferencing system using VoIP technology. Users are reporting issues with video quality, including pixelation, freezing, and echo. To address these problems, you are tasked with conducting a comprehensive assessment of VoIP video quality.

Tasks:

1. Define key video quality metrics relevant to VoIP.
2. Develop test plans to measure video quality under various network conditions.
3. Utilize appropriate testing tools and methodologies.
4. Analyse test results to identify factors impacting video quality.
5. Implement corrective actions to enhance video call experience.

Indicative content 5.5: Testing of VoIP Routing Connectivity

Duration: 1 hr

Theoretical Activity 5.5.1: Description of connectivity testing tools

Tasks:

Task 1: Answer the following questions related to the description connectivity testing tools.

i. What is connectivity testing tool?

ii. What is the connectivity testing tools you know to use?

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to trainer and whole class.

Task 4: In addition, ask questions where necessary.

Task 5: For more clarification, read the **key readings 5.5.1.**

Key readings 5.5.1: Description of connectivity testing tools

- **Ping Command:** Ping and jitter are measures of the speed at which you can request and receive data (ping) and the variation in that response time (jitter). In essence, t hey are measures of the quality of your connection and are used to diagnose perfor mance of real-time applications like video streaming or voice over internet (VoIP).
- **Traceroute (tr) command:** A traceroute can help you troubleshoot your VoIP conne ction. A traceroute maps how internet data travels from your computer to your inte nded destination. Along the way, the traceroute documents the intermediary devic es and measures how long it takes for data to travel from your computer to each po int.

- **Use of connectivity testing tools**

Connectivity testing tools are essential for verifying the path that VoIP calls take through a network. They help identify potential bottlenecks, packet loss, and latency issues that can degrade call quality.

✓ **Key Tools and Their Applications:**

- **Ping:**
  - Verifies network reachability between endpoints.
  - Measures round-trip time (RTT) to assess latency.
  - Can be used to identify network congestion or packet loss.

- **Traceroute:**
  - Maps the path a packet takes to reach a destination.
  - Helps identify network hops experiencing high latency or packet loss.
  - Can be used to troubleshoot routing issues.
- **MTR (MTR):**
  - Combines ping and traceroute functionalities.
  - Provides real-time monitoring of network conditions.
  - Helps identify fluctuating network performance.
- **Network analyzers (Wireshark, tcpdump):**
  - Captures and analyses network traffic.
  - Helps identify VoIP-specific issues like SIP signalling problems, codec issues, and jitter.
  - Can be used to troubleshoot call setup failures and audio quality problems.

✓ **Testing VoIP Routing with Connectivity Tools:**

- **Verify network connectivity:** Ensure endpoints can communicate with each other using ping and traceroute.
- **Check for packet loss and latency:** Use MTR to identify network segments with high packet loss or latency.
- **Analyse VoIP traffic:** Use network analysers to examine SIP signalling, RTP streams, and codec information.
- **Correlate with call quality issues:** Compare test results with call quality reports to identify potential root causes.
- **Optimize network configuration:** Adjust QoS settings, router configurations, or network topology based on test findings.

Points to Remember

- **Testing VoIP Routing with Connectivity Tools:**

✓ **Verify network connectivity:** Ensure endpoints can communicate with each other using ping and traceroute.
✓ **Check for packet loss and latency:** Use MTR to identify network segments with high packet loss or latency.
✓ **Analyse VoIP traffic:** Use network analysers to examine SIP signalling, RTP streams, and codec information.
✓ **Correlate with call quality issues:** Compare test results with call quality reports to identify potential root causes.
✓ **Optimize network configuration:** Adjust QoS settings, router configurations, or network topology based on test findings.

Application of learning 5.5.

AAA organization has recently expanded its VoIP network to accommodate new office locations. To ensure optimal performance and identify potential issues, you are tasked with conducting a comprehensive assessment of VoIP routing connectivity, generating detailed test reports, and analysing the results to recommend improvements.

Tasks:

1. Conduct in-depth testing of VoIP routing paths between different network locations.
2. Utilize appropriate connectivity testing tools to measure network performance metrics.
3. Analyse test results to identify network bottlenecks, latency, and packet loss issues.
4. Develop a structured VoIP testing report format.
5. Create detailed test reports including test objectives, methodology, results, and recommendations.
6. Communicate test findings and recommendations effectively to relevant stakeholders.

Indicative content 5.6: Elaboration of VoIP Testing Report

Duration: 1 hr

Theoretical Activity 5.6.1: Elaboration of VoIP Testing report

Tasks:

Task 1: Answer the following questions related to the description connectivity testing tools.

i. What is testing report?

ii. Identify the testing report elements.

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to trainer and whole class.

Task 4: In addition, ask questions where necessary.

Task 5: For more clarification, read the **key readings 5.6.1.**

Key readings 5.6.1.: Elaboration of VoIP Testing report

- **Understanding the Core of Your VoIP Testing Report**

A VoIP testing report is a comprehensive document that evaluates the performance and quality of a Voice over Internet Protocol system. It's crucial for businesses and service providers to understand the intricacies of this report to make informed decisions about their VoIP infrastructure.

- **Key Components of a VoIP Testing Report**

A typical VoIP testing report covers a wide range of parameters. Here's a breakdown of essential elements and their implications:

✓ **Call Quality Metrics**

- **Mean Opinion Score (MOS):** This metric measures the perceived quality of audio by human listeners. A higher MOS indicates better call quality.
- **Packet Loss:** The percentage of data packets that fail to reach their destination. High packet loss leads to audio interruptions and poor call quality.
- **Jitter:** Variations in packet arrival times, which can cause audio distortion. Low jitter is essential for clear audio.
- **Latency:** The time it takes for data packets to travel from one endpoint to another. High latency results in echo and delay in conversations.

- **One-Way Delay:** The time it takes for audio to travel from one endpoint to another. Excessive one-way delay can affect call synchronization.

✓ **Call Setup and Termination**

- **Call Setup Time:** The time taken to establish a call connection. Shorter setup times enhance user experience.
- **Call Drop Rate:** The percentage of calls that are abruptly terminated before completion. A high drop rate indicates network or system issues.
- **Call Blocking Rate:** The percentage of calls that cannot be completed due to network congestion or other reasons.

✓ **Network Performance**

- **Bandwidth Utilization:** The amount of network bandwidth consumed by VoIP traffic. It's essential to monitor bandwidth usage to avoid congestion.
- **Network Congestion:** Occurs when network traffic exceeds capacity, leading to performance degradation.
- **Quality of Service (QoS):** Evaluates how well the network prioritizes VoIP traffic over other applications.

✓ **Feature Testing**

- **Call Features:** Verification of features like call transfer, conference calling, voicemail, and others.
- **Device Compatibility:** Assessment of VoIP phone compatibility and performance.
- **Interoperability:** Testing compatibility with different VoIP platforms and networks.

✓ **Security**

- **Vulnerability Assessment:** Identification of potential security threats to the VoIP system.
- **Encryption:** Evaluation of encryption protocols used to protect call data.
- **Authentication:** Verification of user authentication mechanisms.

✓ **Interpreting the Results**

Once you have a clear understanding of these metrics, you can interpret the results to identify areas of improvement. For example:

- **Low MOS, high packet loss, and jitter:** Indicate network issues that need to be addressed.
- **High call drop rate and blocking rate:** Suggest capacity problems or network congestion.
- **Slow call setup time:** May point to issues with VoIP servers or network configuration.

✓ **Taking Action**

Based on the report findings, you can implement specific actions to enhance VoIP performance. These may include:

- Upgrading network infrastructure
- Optimizing network configuration
- Implementing QoS mechanisms
- Improving VoIP server performance
- Enhancing security measures

✓ **Additional Considerations**

- **Test Environment:** The accuracy of the report depends on the test environment. It should closely resemble real-world conditions.
- **Testing Methodology:** The testing methodology used should be standardized and repeatable.
- **Regular Testing:** VoIP systems should be tested regularly to identify and address issues proactively.

✓ **Description of testing cycle**

It is a defined period consisting of a start date, an end date, and a list of manual testers. Create testing cycles to plan and execute specific sets of manual tests. The Testing Cycles area lists all testing cycles of the current project.

✓ **Description of testing techniques:**

Testing techniques are the best practices used by the testing team to assess the developed software in regards to given requirements. These techniques ensure the overall quality of the product or software including performance, security, customer experience, and so on.

Points to Remember

- ✓ **Description of testing cycle** is a defined period consisting of a start date, an end date, and a list of manual testers. Create testing cycles to plan and execute specific sets of manual tests. The Testing Cycles area lists all testing cycles of the current project.
- ✓ **Description of testing techniques**: Testing techniques are the best practices used by the testing team to assess the developed software in regards to given requirements. These techniques ensure the overall quality of the product or software including performance, security, customer experience, and so on.

Indicative content 5.7: Description of Testing Report Elements

Duration: 1 hr

Theoretical Activity 5.7.1: Description of testing report elements

Tasks:

Task 1: Answer the following questions related to the description connectivity testing tools.

i. Describe the use of testing report.

ii. Outline the testing report elements.

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to trainer and whole class

Task 4: In addition, ask questions where necessary.

Task 5: For more clarification, read the **key readings 5.7.1.**

Key readings 5.7.1.: Description of testing report elements

A VoIP system testing report is a comprehensive document that evaluates the performance and quality of a Voice over Internet Protocol system.

It typically includes the following elements:

1. Executive Summary

- ✓ Brief overview of the testing objectives
- ✓ Summary of key findings and recommendations
- ✓ High-level assessment of VoIP system performance

2. Test Methodology

- ✓ Detailed description of the testing environment (hardware, software, network configuration)
- ✓ Test cases and scenarios used.
- ✓ Testing tools and equipment employed.
- ✓ Performance metrics and measurement methods

3. Call Quality Metrics

- ✓ **Mean Opinion Score (MOS):** Subjective assessment of call quality.
- ✓ **Packet Loss:** Percentage of data packets lost during transmission.
- ✓ **Jitter:** Variation in packet arrival times
- ✓ **Latency:** Time delay in network communication
- ✓ **One-Way Delay:** Time taken for audio to travel from one endpoint to another.
- ✓ **Echo Cancellation:** Effectiveness of echo suppression
- ✓ **Noise Reduction:** Performance of noise filtering

4. Call Setup and Termination

- ✓ Call setup time
- ✓ Call drop rate
- ✓ Call blocking rate
- ✓ Call completion rate
- ✓ Call hold and resume functionality

5. Network Performance

- ✓ Bandwidth utilization
- ✓ Network congestion
- ✓ Quality of Service (QoS) evaluation
- ✓ Network jitter and latency measurements

6. Feature Testing

- ✓ Functionality of call features (transfer, conference, voicemail, etc.)
- ✓ Device compatibility (phones, softphones, adapters)
- ✓ Interoperability with other systems (PBX, SIP servers)

7. Security

- ✓ Vulnerability assessment
- ✓ Encryption strength and implementation
- ✓ Authentication mechanisms
- ✓ Security policy compliance

8. Performance Under Stress

- ✓ System behaviour under heavy load conditions
- ✓ Call quality degradation under stress
- ✓ Resource utilization (CPU, memory, network)

9. Test Results and Analysis

- ✓ Detailed presentation of test results with graphs and charts
- ✓ Correlation of test results with performance metrics
- ✓ Identification of performance bottlenecks and areas for improvement
- ✓ Comparison with performance benchmarks (if applicable)

10. Recommendations

- ✓ Suggested actions to improve VoIP system performance.
- ✓ Prioritization of recommendations based on impact.
- ✓ Cost-benefit analysis of proposed solutions

11. Appendices

- ✓ Raw test data
- ✓ Detailed test procedures
- ✓ Configuration files

- ✓ Additional supporting documentation

- **Project information**

Test Report is a document which contains a summary of all test activities and final test results of a testing project.
Each test report contains a report header with: The version of Test Real-time used to generate the test as well as the date of the test report generation. The path and name of the project files used to generate the test. Among the project information there must be Name and Description.

- **Test objective**

The objective of test reports is to analyse software quality and provide valuable information in the form of references and feedback for quick decision-making. It showcases the tester's point of view on the testing project and informs stakeholders about the status and possible risks.

- ✓ **Test types**

There are three basic types of test case reporting in software testing:

- Test incident report.
- Test cycle report.
- Test summary report.

- ✓ **Purpose**

A test report is an organized summary of testing objectives, activities, and results. It is created and used to help stakeholders (product manager, analysts, testing team, and developers) understand product quality and decide whether a product, feature, or a defect resolution is on track for release.

- **Test summary**

A test report summary contains all the details of the testing process, what was tested, when was it tested, how it was tested, and the environments where it was tested. A test summary shows if test is passed, failed or blocked.

- **Defect: What would you include in a report about a defect?**

A defect report is a document that includes complete details about the application/software defects, sources, what are the actions needed to resolve them, and the expected result. Developers can check this defect report, understand the problem & its solution and make improvements accordingly. defect report contains description, priority and status in details.

- **Use of testing report format**

Test Report is a document which contains a summary of all test activities and final test results of a testing project. Test report is an assessment of how well the Testing is performed. Based on the test report, stakeholders can evaluate the quality of the tested product and decide on the software release.

Points to Remember

- **Project information:** Test Report is a document which contains a summary of all test activities and final test results of a testing project.
- **Test objective:** The objective of test reports is to analyse software quality and provide valuable information in the form of references and feedback for quick decision-making.

Learning outcome 5 end assessment

Theoretical assessment

Q1. Choose the correct answer.

i. Which of the following is NOT a primary category of VoIP testing tools? Choose one

a) Call quality testing tools
b) Network performance monitoring tools
c) Security testing tools
d) Hardware testing tools

ii Open-source tools can be effective for VoIP testing but may lack comprehensive features compared to commercial options.

a) True
b) False

iii. Which metric measures the perceived quality of a voice call on a scale of 1 to 5?

a) Packet Loss
b) Mean Opinion Score (MOS)
c) Jitter
d) Latency

iv Subjective testing involves using technical equipment to measure audio quality metrics.

a) True
b) False

v. Which of the following is NOT a common audio quality issue in VoIP calls?

a) Echo
b) Noise
c) Latency
d) Pixelation

vi Which metric measures the smoothness of video playback?

a) Frame rate
b) Bitrate
c) Packet loss
d) Latency

vii High jitter can cause video to appear pixelated or blocky.

a) True

b) False

viii Which tool is commonly used to capture network traffic for analysis?

a) Wireshark
b) VoIP tester
c) Audio quality analyser
d) Video conferencing software

ix. Measuring latency between network endpoints is crucial for identifying VoIP routing issues.

a) True
b) False

x. Which section of a VoIP testing report should provide an overview of the testing objectives and methodology?

a) Introduction
b) Test results
c) Recommendations
d) Appendices

xi. A VoIP testing report should include screenshots of test results for better visualization.

a) True
b) False

Q2. I) Match the metric with its description.

SN	Metrics	Descriptions
1	a) Jitter	i. Loss of data packets during transmission
2	b) Latency	ii. Perceived quality of audio or video
3	c) Packet Loss	iii. Variation in packet arrival time
4	d) Bitrate	iv. Delay in packet delivery
5	e) MOS	v. Amount of data transmitted per second

Q2. II) Match the report section with its content.

SN	Report	Contents
1	a) Executive summary	iv. Detailed description of testing procedures and tools
2	b) Test methodology	v. Concise overview of key findings and recommendations
3	c) Test results	vi. Graphical representations of data and performance metrics
4	d) Recommendations	vii. In-depth analysis of test data and identified issues

5	e) Appendices	viii.	Supporting documentation, raw data, and additional information

Practical assessment

CCC organization has recently implemented a new VoIP system to replace its traditional analogy phone system. To ensure optimal performance, reliability, and user satisfaction, you are tasked with conducting a comprehensive test of the VoIP system.

Tasks:

1. Develop a comprehensive test plan covering various aspects of VoIP functionality.
2. Execute test cases to verify system performance and identify potential issues.
3. Measure key performance indicators (KPIs) such as call quality, latency, jitter, and packet loss.
4. Analyse test results to identify areas for improvement.
5. Generate detailed test reports documenting test procedures, results, and recommendations.

END

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Learning Outcome 6: Maintain VoIP System.

Indicative contents

6.1. Setting of backups schedule
6.2. Checking hardware and software functionalities
6.3. Checking VoIP protection measures
6.4. Application of VoIP corrective maintenance measures
6.5. Upgrading VoIP system

Key Competencies for Learning Outcome 6: Maintain VoIP System.

Knowledge	Skills	Attitudes
• Description of setting of backup schedule • Description of hardware and software functionalities • Description of VoIP protection measures • Application of VoIP corrective maintenance measures • Description of diagnosing the VoIP system Problem • Description of VOIP system diagnosis procedures • Description of VoIP system upgrading	• Checking hardware and software functionalities • Checking VoIP protection measures • Applying VoIP corrective maintenance measures • Diagnosing the VoIP system problem • Upgrading VoIP system	• Being vigilance towards security threats. • Having commitment to data protection. • Having accountability for system performance. • Being Patience and persistence in issue resolution. • Being Supportive and service-oriented

Duration: 10 hrs

Learning outcome 6 objectives:

By the end of the learning outcome, the trainees will be able to:

1. Describe properly testing of backup schedule according to the Client's requirements.
2. Describe properly hardware and software functionalities based on VoIP system requirements.
3. Check correctly hardware and software functionalities based on VoIP system requirements.
4. Describe properly VoIP protection measures according to the VoIP system security requirements.
5. Check properly VoIP protection measures according to the VoIP system security requirements.
6. Apply VoIP corrective maintenance measures based on VoIP problem identified.
7. Describe properly Diagnosing of VoIP system problem based on VoIP problem identified.
8. Apply properly VoIP system problem based on VoIP problem identified.
9. Describe properly VoIP system according to the VoIP system functionality.
10. Upgrade correctly VoIP system according to the VoIP system functionality.

Resources

Equipment	Tools	Materials
● Channel bank ● IP phone ● Power backup (UPS) ● IP PBX Server/PBX Server VoIP Router ● VoIP Gateway ● VoIP Headset ● PoE Switch ● Computer	● VoIP Software tools (ZDNet Broadband Speed Test, 8×8 VoIP Test, Speed Test, Dotcom-Monitor, SolarWinds, VoIP Spear...)	● Internet bundles

Indicative content 6.1: Setting of Backups Schedule.

Duration: 2 hrs

Theoretical Activity 6.1.1: Description of Disaster recovery, starting time and completion time

Tasks:

Task 1: Answer the following questions related to the description of disaster recovery.

i. What is disaster recovery?
ii. What are the recovery objectives?
iii. Step down how to restore the files that are deleted.

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to trainer and whole class.

Task 4: In addition, ask questions where necessary.

Task 5: For more clarification, read the **key readings 6.1.1**

Key readings 6.1.1: Description of Disaster recovery, starting time and completion time.

Setting of backups schedule

Setting up a backup schedule for your VoIP (Voice over Internet Protocol) system is crucial to ensure business continuity and data protection. Regular backups help you recover quickly from unexpected events like hardware failures, data corruption, or disasters.

- **Disaster recovery**

VoIP disaster recovery is a crucial part of business continuity planning that focuses on ensuring that voice communication services can be rapidly restored in the event of disasters or system failures.

Components of VoIP Disaster Recovery:

1. **Risk Assessment:**

- Identify potential risks and threats to the VoIP system, such as hardware failures, network outages, natural disasters, and cyberattacks.
- Evaluate their potential impact on voice communications.

2. **Business Impact Analysis (BIA):**

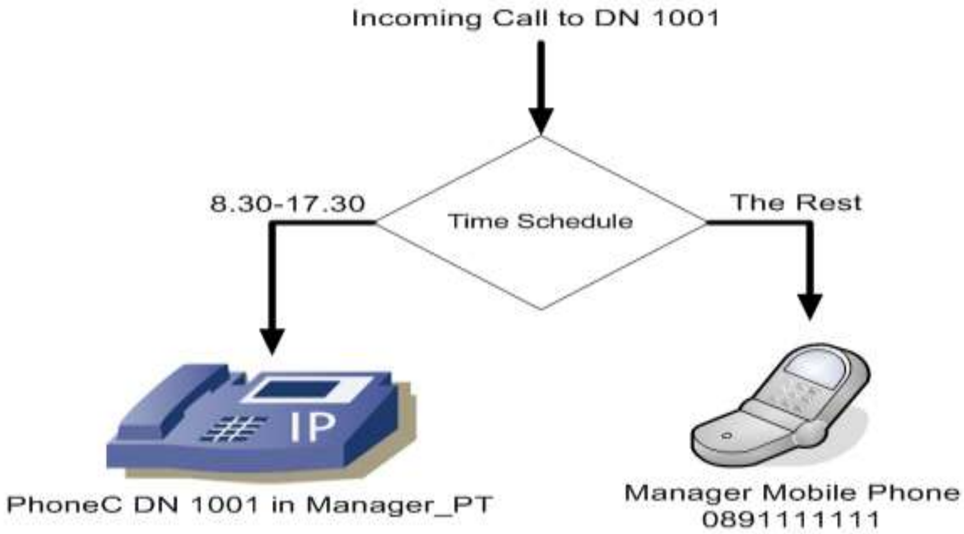
- Determine the criticality of voice communications to business operations.
- Understand the financial, operational, and legal implications of VoIP system disruptions.

3. **Recovery Objectives:**

- Establish Recovery Time Objectives (RTO) and Recovery Point Objectives (RPO).
- RTO defines how quickly VoIP services need to be restored.
- RPO determines the acceptable data loss in case of a disaster.

- **Starting time and completion time**

It's essential to determine both the starting time and the completion time for each backup job. The timing of backups can impact your network's performance and ensure that the backup process does not interfere with normal operations.



1. **Starting Time:**

- **Off-Peak Hours:** Schedule backups during off-peak hours when network traffic is lowest. This reduces the chances of backup processes causing network congestion and negatively impacting call quality during business hours.
- **Late Evening or Early Morning:** Many organizations choose to initiate backups late in the evening or early in the morning when employees are not actively using the network. This minimizes disruption to regular business activities.
- **Consider Time Zones:** If your organization operates in multiple time zones, schedule backups according to the local time of the affected sites to ensure that they occur during off-peak hours at each location.
- **Dependency on Data Updates:** Schedule backups after data updates have been completed. For example, if you update call routing configurations daily, ensure that backups occur after these updates to capture the most current data.
- **Consider Maintenance Windows:** If your organization has scheduled maintenance windows, coordinate backup times with these maintenance periods to minimize network disruptions.

2. **Completion Time:**

- **Allow for Sufficient Time:** Ensure that backups have enough time to complete without interfering with the start of business operations or network usage. The completion time should be well before peak network activity.
- **Monitoring and Verification:** Plan for a verification process after backups complete to confirm that they were successful. This step may add extra time to the completion phase.

- **Notifications:** If any issues are detected during the backup process, set up alerts and notifications to promptly address them. This may extend the completion time, but it's crucial for proactive problem resolution.
- **Backup Retention:** If backups involve archiving data for long-term retention, factor in the time needed to move backup files to appropriate storage locations.
- **Adjustment for Backup Size:** The size of your backups can affect the completion time. Larger backups may take longer, so schedule accordingly.
- **Parallel Backups:** If your VoIP system is extensive, consider parallel backups where multiple components are backed up simultaneously. This can reduce the overall time required.
- **Overlapping Backup Windows:** If your organization operates in multiple time zones, consider overlapping backup windows for different sites to optimize backup completion times.

- **Restore files that are accidentally deleted.**

Accidental file deletion can occur in any IT system, including VoIP environments. Whether it's call recordings, configuration files, or critical data, knowing how to effectively restore accidentally deleted files is crucial for maintaining VoIP system integrity.

What you must consider:

- **Identify the Deleted Files:** Begin by determining which files have been accidentally deleted. This may include configuration files, call recordings, or other data essential to the VoIP system.
- **Determine Backup Availability:** Check if there are backups of the deleted files. VoIP systems often include backup and recovery mechanisms, so explore your available backup options.

- **Locate and Access Backups:** Access your backup system or storage location to retrieve the deleted files. This might involve restoring from on-premises backups, cloud-based backups, or even backups from secondary systems.
- **Verify Data Integrity:** Before restoring the files, verify the integrity of the backup data to ensure that it contains the most recent and complete version of the deleted files.
- **Restore from Backup:** Use the appropriate tools and procedures to restore the deleted files from the backup to their original location or an alternate location.
- **Test Restored Files:** After restoration, test the recovered files to ensure they function as expected. For call recordings, check the audio quality and completeness.
- **Document the Restoration:** Keep records of the restoration process, including the date, time, files restored, and any issues encountered. This documentation can be valuable for future reference.
- **Prevention Measures:** Implement preventive measures to reduce the risk of future accidental file deletions, such as setting file permissions and access controls to limit who can delete files.
- **Regular Backups:** Emphasize the importance of regularly backing up VoIP system data. Frequent backups help ensure that the most recent data is available for recovery.

Accidental file deletions can disrupt VoIP operations, but with a well-structured backup and recovery plan, along with preventive measures, organizations can effectively restore deleted files and minimize downtime and data loss.

- **Effect of backup activities on production activities**

Backup activities in a VoIP (Voice over Internet Protocol) environment are essential for data protection and disaster recovery. However, these activities can have both positive and negative effects on production activities. It's crucial to strike a balance between data protection and minimizing disruption to voice communication services.

- ✓ **Positive Effects:**

- **Data Protection:** Backup activities help safeguard critical data, including call records, configuration files, and call recordings. This protection ensures that data remains intact even in the event of system failures or data loss.
- **Disaster Recovery:** Backups provide a safety net in case of disasters, such as hardware failures, network outages, or cyberattacks. They allow for swift recovery, reducing downtime and maintaining communication capabilities.
- **Data Integrity:** Regular backups help maintain data integrity by preserving the most recent and complete information. This ensures that call records are accurate and that call recordings are accessible for compliance and auditing purposes.
- **Peace of Mind:** Knowing that data is regularly backed up can provide peace of mind to both IT administrators and users, as they are aware that their voice communications are protected against data loss.

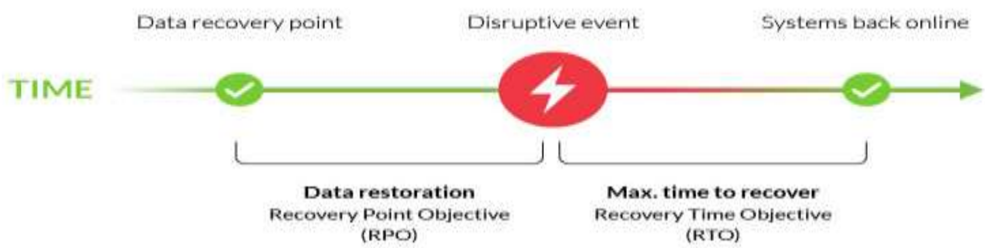
- ✓ **Negative Effects:**

- **Network Overhead:** During backup activities, network resources may be consumed, potentially causing network congestion. This can result in reduced call quality or latency, affecting ongoing voice communications.
- **Downtime During Backups:** Depending on the backup strategy and network capacity, VoIP services may need to be temporarily interrupted during the backup process, causing downtime for users.
- **Resource Utilization:** Backup activities can consume server resources and storage space, potentially impacting the performance of the VoIP servers and the servers used for backup.

- **Complexity and Maintenance:** Managing backup activities, including scheduling, monitoring, and ensuring the integrity of backup data, adds complexity to the network infrastructure and requires ongoing maintenance.

- **Compliance with recovery time and recovery point objectives**

Compliance with Recovery Time Objectives (RTO) and Recovery Point Objectives (RPO) is essential for effective disaster recovery in VoIP (Voice over Internet Protocol) systems. These objectives determine the maximum acceptable downtime and data loss during a disaster or system failure.



✓ **Recovery Time Objective (RTO):**

- RTO is the maximum amount of time a VoIP system can be down before it negatively impacts business operations.
- In the context of VoIP, RTO means the time it takes to restore voice communication services after a disaster or system failure.
- Compliance with RTO involves ensuring that the VoIP system can be quickly recovered within the specified timeframe to minimize downtime and maintain call availability.

✓ **Recovery Point Objective (RPO):**

- RPO is the maximum amount of data loss that is acceptable during a disaster or system failure.
- In VoIP systems, RPO indicates how much call data (e.g., call records, voicemail messages) can be lost without causing significant disruptions.
- Compliance with RPO means that the VoIP system's data backups and recovery processes are designed to minimize data loss and recover data up to the specified point in time.

Points to Remember

Setting a backup schedule for a VoIP system involves determining the frequency and method for backing up critical VoIP system data and configurations. This ensures business continuity and enables quick system restoration in case of failures, disasters, or security breaches.

Application of learning 6.1.

Your organization relies heavily on its VoIP system for daily operations. To ensure business continuity and minimize disruptions due to system failures or disasters, you are tasked with establishing a robust backup schedule for the VoIP system.

Tasks:

1. Identify critical VoIP system components to be backed up.
2. Determine the optimal backup frequency based on data criticality.
3. Select appropriate backup methods (full, incremental, differential).
4. Establish a backup retention policy to manage storage space.
5. Implement backup procedures and test their effectiveness.
6. Develop a disaster recovery plan incorporating the backup schedule.

Indicative content 6.2: Checking Hardware and Software Functionalities.

Duration: 2 hrs

Theoretical Activity 6.2.1: Description of hardware and software functionalities.

Tasks:

Task 1: In small groups, you are requested to answer the following questions related to the description of VoIP system hardware and software functionalities.

- **i.** What is the meaning of functionality checking?
- ii. Describe the process of conducting functional testing on a VoIP phone, including the test parameters and expected outcomes.
- iii. How can you effectively test the compatibility between different VoIP hardware and software components to ensure optimal system performance?

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to trainer and whole class

Task 4: In addition, ask questions where necessary.

Task 5: For more clarification, read the **key readings 6.2.1**

Key readings 6.2.1: Description of hardware and software functionalities

Checking the hardware and software functionalities of a VoIP (Voice over Internet Protocol) system is essential to ensure that it operates smoothly and delivers high-quality voice communication services. Regular assessments help identify and address issues, optimize performance, and maintain reliable service.

- **Hardware**

To check the hardware functionality of a VoIP (Voice over Internet Protocol) system, you should conduct a thorough assessment of the physical components and infrastructure to ensure they are in good working condition and capable of supporting VoIP services.

- ✓ **Network Infrastructure:**

Examine the overall network infrastructure, including routers, switches, and cabling, to ensure that they are in good condition and properly configured for VoIP traffic.

- **VoIP Phones**

Verify that VoIP phones are powered on and connected to the network.

Test making and receiving calls on different phones.
Check for any display, keypad, or button issues on the phones.

- **Routers and Switches**

Inspect routers and switches for physical damage and loose connections.
Ensure that Power over Ethernet (PoE) is properly delivering power to connected devices.
Validate that VLAN and Quality of Service (QoS) settings are appropriately configured to prioritize voice traffic.

- **Gateways and Analog Interfaces**

Check the condition of gateways and analogy interfaces for any visible issues.
Test analogy connections for voice quality and functionality.

- **Server Hardware**

Verify the health of server hardware hosting VoIP applications or PBX (Private Branch Exchange) systems.
Ensure that servers are running within acceptable temperature ranges and are free from hardware faults.

- **Uninterruptible Power Supplies (UPS)**

Confirm that UPS systems are operational and capable of providing backup power during electrical outages.

- **Cabling**

Inspect Ethernet cables and connectors for damage or wear.
Verify that cables are securely connected to devices without signal interference.

- **Network Switch Ports**

Check that network switch ports are correctly configured and that VLANs and QoS settings are optimized for VoIP traffic.

- ✓ **Performance**

Evaluating and optimizing the performance of a VoIP (Voice over Internet Protocol) system is crucial to ensure high-quality voice communication. VoIP performance directly impacts call quality, user satisfaction, and the overall functionality of the system.

- **Bandwidth Assessment**

Determine the required bandwidth for your VoIP system, considering factors such as the number of concurrent calls, codec selection, and data traffic. Ensure that your network has sufficient bandwidth to support VoIP without congestion.

- **Jitter Control**

Address network jitter, which can disrupt voice quality. Implement adaptive jitter buffers, network prioritization, or dynamic jitter control methods to compensate for varying packet arrival times.

- **Redundancy and Failover**

Implement redundancy and failover mechanisms to ensure that your VoIP system continues to operate in the event of network or hardware failures. Redundant servers, network links, and SIP trunks can enhance system resilience.

- **Load Testing**

Conduct load testing to evaluate the system's capacity to handle many simultaneous calls and identify performance limitations.

- ✓ **Status**

Checking the status of hardware and software functionalities of a VoIP (Voice over Internet Protocol) system is essential to ensure the system's reliability and quality of service. Regular monitoring and status checks help identify and address issues promptly.

- **Physical Inspection**

Physically inspect VoIP hardware components, such as phones, routers, switches, gateways, and servers, to ensure they are in good condition and securely connected.

- **Power and Connectivity**

Verify that all hardware devices are powered on and properly connected to power sources and the network.

- **Cabling and Connectors**

Check Ethernet cables and connectors for any signs of wear and tear or loose connections. Ensure they are securely connected.

- **Software**

There are several software components and applications that make up a VoIP (Voice over Internet Protocol) system. These software elements work together to enable voice communication over the internet.

- **Softphones**

Softphones are software-based applications that allow users to make and receive VoIP calls on their computers, smartphones, or tablets. Examples of popular softphone applications include Zoiper, X-Lite, and Bria.

- **PBX Software**

PBX (Private Branch Exchange) software manages call routing, voicemail, call conferencing, and other features within an organization. Open-source PBX solutions like Asterisk and FreeSWITCH are widely used. Commercial options include 3CX and Cisco Unified Communications Manager.

- **VoIP Clients**

VoIP clients are applications used to connect to VoIP services. This category includes both softphones and hardware-based VoIP phones that run specific VoIP client software.

- ✓ **Status**

- **Network Status**

Check the status of your network infrastructure, including routers, switches, firewalls, and Internet Service Providers (ISPs), to ensure they are operational and properly configured for VoIP traffic.

- **Call Quality**

Monitor the quality of VoIP calls, assessing audio clarity, voice distortions, echo, latency, jitter, and packet loss. Real-time call quality metrics are crucial.

- ✓ **Updates**

Keeping your VoIP (Voice over Internet Protocol) system up to date is essential to ensure the security, performance, and functionality of the system. Regular updates should cover both hardware and software components.

- **Firmware Updates**

Update the firmware of VoIP devices, such as IP phones and gateways, to ensure they have the latest security patches, bug fixes, and feature enhancements.

- **Software Updates**

Keep your VoIP software, including PBX software, softphone applications, and any other VoIP-related software, up to date. Updates often include security patches and new features.

- **Operating System Updates**

If your VoIP system runs on a server with an operating system, make sure the operating system is regularly updated to address security vulnerabilities and improve system stability.

Security Updates

Stay vigilant about security updates and patches for your VoIP system. Security vulnerabilities in VoIP systems can be targeted by attackers, so ensure that your system is protected with the latest security updates.

✓ **Services of features**

VoIP (Voice over Internet Protocol) systems offer a wide range of services and features to enhance voice communication. These services and features are designed to improve call quality, flexibility, and productivity.

Voice Calls

Basic VoIP functionality includes making and receiving voice calls over the internet.

Video Calls

Many VoIP systems support video calls, allowing users to have face-to-face video conversations.

Call Waiting

Users can receive notifications and switch between incoming calls without disconnecting the current call.

Call Forwarding

Calls can be redirected to another number or device, such as a mobile phone or another extension.

Call Transfer

Users can transfer calls to other extensions or users, facilitating efficient call routing within an organization.

Points to Remember

- Regularly assessing the performance and capabilities of VoIP system components is c rucial for maintaining optimal system operation. This involves evaluating hardware s uch as IP phones, gateways, and servers, as well as software including IP PBX, VoIP ap plications, and network management tools.
- Functional, performance, compatibility, and security testing should be conducted to i dentify and address potential issues. By ensuring that all hardware and software com ponents work as intended, organizations can enhance call quality, reliability, and use r satisfaction.

Application of learning 6.2.

Your organization has been experiencing intermittent issues with its VoIP system, including call quality problems, dropped calls, and system slowdowns. To identify the root cause of these problems and ensure optimal system performance, you are tasked with conducting a comprehensive check of the VoIP system's hardware and software functionalities.

Tasks:

1. Identify key components of the VoIP system for evaluation.
2. Develop a testing methodology to assess hardware and software performance.
3. Conduct functional tests to verify system capabilities.
4. Analyse test results to identify potential issues.
5. Document findings and recommendations for system improvement.

Indicative content 6.3: Checking VoIP Protection Measures.

Duration: 2 hrs

Theoretical Activity 6.3.1: Description of VoIP protection measures.

Tasks:

Task 1: Answer the following questions related to the description of VoIP system upgrading.

i. What do you understand by VoIP protection measures.

ii. How to check VoIP system?

iii. Differentiate physical security from logical security.

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to trainer and whole class.

Task 4: In addition, ask questions where necessary.

Task 5: For more clarification, read the **key readings 6.3.1**

Key readings 6.3.1: Description of VoIP protection measures.

Protecting your VoIP (Voice over Internet Protocol) system from security threats and ensuring the privacy and integrity of voice communications is essential. you can significantly enhance the security and reliability of your VoIP system and reduce the risk of security breaches or disruptions in voice communication services. Regularly review and update your security measures to adapt to evolving threats and vulnerabilities.

- **Physical security**

Physical security is a critical aspect of protecting your VoIP (Voice over Internet Protocol) system, as it involves safeguarding the physical infrastructure and equipment that make up your VoIP network. To ensure the physical security of your VoIP system

✓ **Access Control**

Restrict physical access to VoIP equipment and server rooms to authorized personnel only. Use electronic card access systems or biometric authentication where appropriate.

✓ **Secure Location**

Place VoIP servers, gateways, and other critical equipment in secure, controlled-access areas. These areas should be locked and monitored for unauthorized entry.

✓ **Alarms and Intrusion Detection**

Install intrusion detection systems and alarms to detect unauthorized access or tampering with VoIP equipment. Ensure that alerts are promptly sent to security personnel.

- ✓ **Physical Hardware Locks**

Use physical locks and locking cabinets or racks to secure VoIP hardware and networking equipment in place. This prevents theft or unauthorized removal.

- ✓ **Cable Management**

Keep network cables organized and hidden to prevent tampering and unauthorized cable connections.

- ✓ **Labelling and Documentation**

Label VoIP equipment and cables to make it easy to identify and maintain them. Maintain up-to-date documentation of equipment inventory and configurations.

- **Logical security**

It is also known as cybersecurity, is the protection of your VoIP (Voice over Internet Protocol) system against digital threats, data breaches, and unauthorized access through logical or software-based means.

- ✓ **Access Control**

Implement strong user authentication methods and access control measures, such as usernames and passwords, two-factor authentication (2FA), and role-based access control (RBAC) to limit system access.

- ✓ **Encryption**

Encrypt signaling and media streams using secure protocols like TLS (Transport Layer Security) and SRTP (Secure Real-time Transport Protocol) to protect call data from interception.

- ✓ **Secure Configuration**

Configure VoIP servers, routers, and firewalls with security in mind, disabling unnecessary services and following industry best practices.

- ✓ **Backup and Disaster Recovery**

Implement backup and disaster recovery plans to ensure the availability of VoIP services in case of system failures or cyberattacks.

Points to Remember

Checking VoIP protection measures involves assessing the security posture of your VoIP system to identify vulnerabilities and implement countermeasures. It's a proactive approach to safeguarding sensitive communications and preventing unauthorized access.

Application of learning 6.3.

Your organization has experienced a recent increase in security incidents, including unauthorized access attempts and potential data breaches related to the VoIP system. To mitigate these risks and protect sensitive information, you are tasked with conducting a comprehensive review of the VoIP system's security measures.

Tasks:

1. Assess the current security posture of the VoIP system.
2. Identify vulnerabilities and potential threats.
3. Evaluate the effectiveness of existing security measures.
4. Develop recommendations to strengthen VoIP security.
5. Implement security enhancements to protect the system and user data.

Indicative content 6.4: Application of VoIP Corrective Maintenance Measures

Duration: 2 hrs

Theoretical Activity 6.4.1: Application of VoIP corrective maintenance measures

Tasks:

Task 1: Answer the following questions:

i. What is VoIP corrective maintenance measures?
ii. What are the common problems occurring on the VoIP system?
iii. What is the measure taken to fix VoIP faults?

Task 2: Provide the answer for the asked questions and write them on papers.
Task 3: Present the findings/answers to the whole class
Task 4: In addition, ask questions where necessary.
Task 5: For more clarification, read the **key readings 6.4.1**

Key readings 6.4.1.: Application of VoIP corrective maintenance measures

Corrective maintenance measures for VoIP (Voice over Internet Protocol) systems are necessary to address and resolve issues, faults, and disruptions in the operation of the system. These measures are applied to restore the system to normal functioning.

- **Diagnose the VoIP system Problem (Identify the problem)**

Diagnosing a VoIP (Voice over Internet Protocol) system problem is the initial step in addressing and resolving issues with the system. Proper diagnosis helps pinpoint the root cause of the problem, making it easier to apply corrective measures.

✓ **VoIP Security Threats**

VoIP (Voice over Internet Protocol) systems are vulnerable to various security threats that can compromise the confidentiality, integrity, and availability of voice communication. It's important to be aware of these threats and take measures to protect your VoIP infrastructure.

- **Eavesdropping:**

Attackers intercept and listen to voice calls, potentially obtaining sensitive information or conversations. Encryption of voice traffic helps mitigate this threat.

- **Caller ID Spoofing:**

Attackers manipulate Caller ID information to impersonate someone else, potentially facilitating fraud, scams, or impersonation attacks.

- **Toll Fraud:**

Attackers gain unauthorized access to the VoIP system and make long-distance or international calls, resulting in significant financial losses.

- **Vishing (Voice Phishing):**

Like email phishing, attackers use voice calls to trick users into revealing sensitive information, such as passwords or credit card details.

- **Man-in-the-Middle (MitM) Attacks:**

Attackers intercept and manipulate voice traffic between two parties, potentially altering or eavesdropping on conversations.

✓ **Connectivity**

Connectivity in the context of VoIP (Voice over Internet Protocol) refers to the ability of VoIP devices, such as IP phones, softphones, gateways, and servers, to communicate and exchange voice data over the network. Reliable and stable connectivity is essential for maintaining high-quality voice communication.

- **Network Infrastructure:**

The network infrastructure, including routers, switches, and cabling, plays a critical role in VoIP connectivity. Ensure that your network components are properly configured and capable of handling VoIP traffic.

- **Internet Connection:**

A stable and high-quality internet connection is crucial for VoIP connectivity. A reliable and sufficient bandwidth is needed to support voice calls without interruptions.

- **VoIP Gateways:**

VoIP gateways connect VoIP networks to traditional PSTN networks. Ensure gateways are correctly configured and provide seamless connectivity.

- **Remote User Connectivity:**

Ensure that remote users have secure and reliable connectivity to the VoIP system, especially if they use softphones or work from off-site locations.

✓ **Performance**

Performance is a crucial aspect of a VoIP (Voice over Internet Protocol) system, as it directly impacts the quality of voice communication and the overall user experience.

- **Latency:**

Latency is the delay in the transmission of voice data over the network. High latency can result in awkward pauses in conversations and impact call quality. It is essential to minimize latency to maintain real-time communication.

- **Packet Loss:**

Packet loss occurs when voice data packets are dropped or not delivered in sequence. Minimizing packet loss is essential for maintaining clear and uninterrupted voice calls. Techniques like Forward Error Correction (FEC) can help mitigate packet loss.

- **Bandwidth:**

Sufficient and stable bandwidth is necessary for VoIP calls. Inadequate bandwidth can lead to call drops, poor call quality, and delays in call setup. Implement Quality of Service (QoS) settings to prioritize VoIP traffic over other network traffic.

✓ **Functionality**

Functionality in the context of VoIP (Voice over Internet Protocol) systems refers to the features and capabilities that enable voice communication and collaboration. VoIP systems are designed to provide a wide range of functionalities to enhance communication, efficiency, and productivity.

- **Voice Calls:**

Basic VoIP functionality includes making and receiving voice calls over the internet.

- **Video Calls:**

Many VoIP systems support video calls, allowing users to have face-to-face video conversations.

- **Call Waiting:**

Users can receive notifications and switch between incoming calls without disconnecting the current call.

- **Call Forwarding:**

Calls can be redirected to another number or device, such as a mobile phone or another extension.

- **Analyse the problem.**

Analysing a problem in the context of a VoIP (Voice over Internet Protocol) system involves a systematic process of investigation and assessment to identify the root cause and nature of the issue.

❖ **Gather Information:**
Begin by collecting as much information as possible about the problem. This may include reports from users, error messages, call details, and any relevant context.

❖ **Define the Problem:**
Clearly define the problem based on the gathered information. It's essential to have a specific and concise description of the issue. For example, "Users are experiencing poor call quality during peak hours."

❖ **Problem Identification:**
Identify whether the issue is isolated to specific users, locations, or devices, or if it affects the entire VoIP system. This helps determine the scope of the problem.

❖ **User Account and Configuration Check:**
Verify that user accounts, including usernames, passwords, and extension numbers, are correctly configured. Incorrect configurations can lead to registration and call issues.

❖ **Network Infrastructure Assessment:**
Examine the network infrastructure, including routers, switches, and firewalls, for any configuration errors or hardware issues that might affect VoIP traffic.

❖ **Configuration Review:**
Evaluate the configuration settings of VoIP devices, SIP servers, and PBX systems to ensure they are set up correctly. Look for misconfigurations that could be causing the problem.

❖ **Log Analysis:**
Review system logs, including call logs, error logs, and event logs, for any error messages or unusual behaviour that might offer clues to the problem's origin.

✓ **Techniques**

Analysing a problem in a VoIP (Voice over Internet Protocol) system involves using various techniques to gather information, identify root causes, and determine effective solutions.

- **Troubleshooting and Diagnostic Tools:**

Use network diagnostic tools, packet analysers (e.g., Wireshark), and VoIP-specific troubleshooting tools to capture and analyse network traffic. These tools can help identify network issues, packet loss, jitter, and latency.

- **Log Analysis:**

Review system logs, including call logs, error logs, and event logs, to identify error messages and patterns related to the problem. Logs can provide valuable information about system behaviour.

- **Traffic Analysis:**

Analyse network traffic patterns to identify congestion, bandwidth issues, or irregularities. Traffic analysis can reveal if network conditions are affecting VoIP performance.

- **Consultation with Experts:**

Engage with VoIP experts, system administrators, or technical support teams to leverage their knowledge and experience in analysing and resolving VoIP issues.

- **Plan for solution**

When planning for a solution to resolve VoIP (Voice over Internet Protocol) system issues, it's essential to consider the methods and tools you will use, as well as any equipment and materials required.

✓ **Tools**

To troubleshoot and diagnose VoIP (Voice over Internet Protocol) problems effectively, you can utilize a variety of tools and software solutions. These tools are essential for monitoring, analysing, and resolving issues related to VoIP systems.

- **Wireshark:**

A powerful packet capture and analysis tool that allows you to monitor network traffic and analyse SIP signalling and RTP voice packets.

- **SIP Analysers:**

Specialized SIP analysers, such as SIPp and SIP Workbench, focus on SIP signalling protocol analysis and debugging.

- **VoIP Monitoring Software:**

Tools like SolarWinds VoIP & Network Quality Manager or PRTG Network Monitor offer comprehensive VoIP monitoring, call quality assessment, and network performance analysis.

- **Ping and Traceroute:**

Standard network diagnostic tools for testing network connectivity and identifying potential issues in network path.

- ✓ **Equipment**

- **IP Phones or VoIP Softphones:**

IP phones are specialized phones designed for VoIP calls. Alternatively, VoIP softphones are software-based applications that turn computers or mobile devices into VoIP endpoints.

- **VoIP PBX System:**

A VoIP Private Branch Exchange (PBX) system serves as the core of your VoIP network, handling call routing, extension management, and voicemail services.

- **VoIP Gateway:**

VoIP gateways interface between VoIP networks and traditional PSTN (Public Switched Telephone Network), allowing calls to be sent and received from standard telephone lines.

- **Network Routers:**

Routers are essential for routing VoIP traffic within your network and connecting it to external networks, such as the internet.

- **Network Switches:**

Ethernet switches are used to create local area networks (LANs) and manage the flow of data between devices within the network.

- ✓ **Materials**

Setting up and maintaining a VoIP (Voice over Internet Protocol) system requires specific materials to ensure its smooth operation.

- **Ethernet Cables:**

High-quality Ethernet cables (Cat5e, Cat6, or Cat6a) are needed to connect VoIP phones, switches, routers, and other network devices. These cables transmit data and power (in the case of PoE) within the network.

- **RJ45 Connectors and Keystone Jacks:**

RJ45 connectors and keystone jacks are used for terminating Ethernet cables. Properly crimped connectors and jacks ensure secure connections.

- **Patch Panels:**

Patch panels are used to organize and manage Ethernet cable connections in server rooms or data centres. They facilitate cable management and allow for easy reconfiguration.

- **Network Racks and Cabinets:**

Network racks and cabinets provide physical infrastructure for housing network equipment, servers, switches, and other hardware in an organized and secure manner.

- **Cable Management Accessories:**

Cable management accessories, such as cable trays, cable ties, and cable labels, help keep cables organized and reduce clutter in network cabinets.

- **Implement solution/Fix the problem.**

Implementing solutions and fixing problems in a VoIP (Voice over Internet Protocol) system is crucial to ensure that voice communication remains reliable and of high quality.

✓ **Repair**

Repairing a VoIP (Voice over Internet Protocol) system involves addressing and resolving hardware or software issues to restore the system's functionality.

- **Identify the Issue:**

Determine the specific problem with the VoIP system. Is it a hardware failure, software malfunction, network issue, or a combination of these?

- **Isolate the Problem:**

Isolate the issue to determine if it affects specific users, devices, or locations, or if it's a system-wide problem.

- **Backup Configuration:**

Before making any changes or repairs, back up the configuration settings of your VoIP system to avoid data loss and facilitate recovery.

- **Troubleshoot Hardware Issues:**

If the problem is related to hardware, such as malfunctioning VoIP phones, switches, or routers, diagnose the specific faulty components.

- **Replace or Repair Hardware:**

If hardware components are identified as the root cause, replace or repair them. This may involve ordering replacement parts or contacting the manufacturer for warranty service.

- **Update Firmware and Software:**

Ensure that all VoIP devices, servers, and software are running the latest firmware and software updates. Software updates can address known bugs and security vulnerabilities.

- **Configuration Verification:**

Review and verify the configuration settings of VoIP devices, routers, and firewalls to correct any misconfigurations that may be causing the problem.

- **Network Troubleshooting:**

Address network-related issues, such as network congestion, improper QoS settings, or firewall rules that may impact VoIP traffic.

- **Codec Assessment:**

Assess the choice of codecs in use and optimize them to balance call quality and bandwidth utilization.

- **Security Enhancements:**

Implement security measures to prevent future security-related problems, such as unauthorized access, intrusion attempts, or breaches.

- **System Testing:**

After repairing or making changes, thoroughly test the VoIP system to ensure that the problem has been resolved.

- **Documentation:**

Maintain detailed records of the repair process, including the identified problem, steps taken to fix it, and any hardware replacements or software updates.

- **User Communication:**

Communicate with affected users to inform them that the problem has been repaired and provide any necessary instructions or changes they should be aware of.

✓ **Replacement**

When a component in your VoIP (Voice over Internet Protocol) system is malfunctioning or beyond repair, you may need to replace the defective hardware or software to ensure the continued operation of your VoIP system.

- **Identify the Faulty Component:**

First, identify and confirm which hardware or software component in your VoIP system is malfunctioning or needs replacement. This could be a malfunctioning VoIP phone, a router, a server, or even a software application.

- **Procure the Replacement Component:**

Once you've identified the faulty component, order a replacement. This may involve contacting the manufacturer or your supplier to purchase a new device, software license, or component.

- **Backup Configuration:**

Before removing or replacing the faulty component, back up the configuration settings of the existing hardware or software. This backup will help you ensure that the new component is configured correctly.

- **Physical Replacement (Hardware):**

If you're replacing hardware, such as a defective VoIP phone or network switch, physically swap out the faulty component with the new one. Ensure that connections are made properly and securely.

- **Installation and Configuration (Software):**

If you're replacing software or a software component, install the new software or component following the manufacturer's instructions. Configure it based on the backup settings and any necessary adjustments.

- **Testing and Validation:**

After replacement, thoroughly test the new component or software to ensure that it functions as expected. Test VoIP calls, network connectivity, and any specific features related to the replaced component.

- **Configuration Restoration:**

If you backed up the configuration settings, restore the configuration from the backup to ensure that the new component operates according to your previous settings.

- **User Communication**:

If the replacement affects users, communicate with them to inform them of the replacement and any changes or instructions they should be aware of. Provide necessary training or guidance as needed.

- ✓ **Update**

Updating your VoIP (Voice over Internet Protocol) system is essential to keep it secure, reliable, and up to date with the latest features and improvements.

- **Identify Components for Update:**

Determine which components of your VoIP system need updating. This may include VoIP phones, PBX software, routers, network switches, and any other relevant equipment or software.

- **Backup Configuration:**

Before proceeding with updates, backup the configuration settings of the components you plan to update. This backup serves as a safety net in case anything goes wrong during or after the update.

- **Identify Available Updates:**

Check with the manufacturers or vendors of your VoIP components to identify available updates. This can include firmware updates for VoIP phones, software updates for PBX systems, and security patches.

- **Review Release Notes:**

Carefully review the release notes or change logs provided by the manufacturers or vendors. These documents detail what the updates address, new features, and any known issues or considerations.

- **Plan the Update Schedule:**

Schedule updates during periods of low call volume or non-peak hours to minimize disruption. Ensure that you have adequate time for testing and troubleshooting if needed.

- **Testing:**

Thoroughly test the updated components to ensure they function correctly. Test VoIP calls, network connectivity, and any specific features related to the updated components.

- **Continuous Monitoring:**

Continuously monitor the VoIP system and relevant metrics to ensure that the updates have not introduced new issues and that the system continues to operate smoothly.

- **User Communication:**

Communicate with users to inform them about the updates, any new features, or changes that may affect their use of the VoIP system.

- **Documentation:**

Maintain detailed records of the update process, including which components were updated, the update versions, testing results, and any issues encountered during the update.

- ✓ **Upgrade**

Upgrading your VoIP (Voice over Internet Protocol) system is a strategic process to improve its features, functionality, and overall performance.

- **Identify the Need for an Upgrade:**

Determine the specific reasons for the upgrade. It could be to enhance call quality, add new features, improve security, or accommodate growing call volume. Clear objectives are essential.

- **Conduct a System Assessment:**

Perform a thorough evaluation of your existing VoIP system to identify its strengths and weaknesses. Pinpoint areas that require improvement and align them with your upgrade goals.

- **Backup Configuration:**

Before proceeding with any upgrades, back up the configuration settings of your current VoIP system. This backup will serve as a safety net in case issues arise during the upgrade.

- **Develop an Upgrade Plan:**

Create a detailed upgrade plan that outlines the steps, timelines, and responsibilities. Consider the impact on users and schedule the upgrade during a time that minimizes disruption.

- **Procure Necessary Hardware and Software:**

Acquire any required hardware or software for the upgrade, such as new VoIP phones, PBX servers, or network equipment.

- **Firmware and Software Updates:**

Update the firmware for VoIP phones, PBX software, and network components to their latest versions. Ensure compatibility among all system components.

- **Testing of the system**

Testing your VoIP (Voice over Internet Protocol) system is a critical step to ensure that it functions as intended, provides high-quality voice communication, and meets your business needs.

- **Preparations:**

Before testing, ensure that all components of your VoIP system are correctly configured, and that all necessary hardware and software are in place. Make sure that your network infrastructure is optimized for VoIP traffic.

- **Test Plan Development:**

Create a detailed test plan that outlines the specific tests you will conduct, including the objectives, scenarios, and expected outcomes. Identify the key performance indicators you'll monitor.

- **Functional Testing:**

Start with functional testing to verify that all basic call-related functions work as expected. Test call setup, call termination, call transfer, and voicemail.

- **Call Quality Testing:**

Assess call quality by making test calls. Pay attention to factors such as audio clarity, latency, jitter, and packet loss. Use tools like Mean Opinion Score (MOS) testing to quantify call quality.

- **Scalability Testing:**

Test the system's ability to handle increased call volume. Determine the system's limitations and its capacity to support additional users and concurrent calls.

- **Network Testing:**

Assess the network's performance, particularly its ability to prioritize VoIP traffic. Evaluate Quality of Service (QoS) settings and ensure that VoIP traffic is not affected by network congestion.

Security Testing:

Perform security testing to identify vulnerabilities and ensure that your VoIP system is protected from unauthorized access and threats. Test for secure call encryption and data protection.

Points to Remember

- ✓ **Diagnose the VoIP System Problem** involves systematically identifying the root cause of issues within a VoIP system. It's the first step towards resolving problems and ensuring optimal system performance.
- ✓ **Common VoIP problems and potential causes include** Poor call quality, dropped calls, Call failures and System outages.

Application of learning 6.4.

AAA Company is experiencing recurring issues with its VoIP system, including frequent call drops, poor audio quality, and system outages. These problems are impacting productivity and user satisfaction. To address these challenges, you are tasked with implementing corrective maintenance measures to restore normal system operation.

Tasks:

1. Identify the root cause of VoIP system failures.
2. Develop and implement corrective actions to resolve issues.
3. Restore normal VoIP system functionality.
4. Document corrective maintenance procedures for future reference.
5. Implement preventive measures to reduce recurrence of issues.

Indicative content 6.5: Upgrading VoIP System.

Duration: 2 hrs

Theoretical Activity 6.5.1: Description of VoIP system upgrading.

Tasks:

Task 1: In small groups, you are requested to answer the following questions related to the description of VoIP system upgrading.

i. What is VoIP system upgrading?

ii. What are the steps followed to upgrade VoIP system?

Task 2: Provide the answer for the asked questions and write them on papers.

Task 3: Present the findings/answers to trainer and whole class.

Task 4: In addition, ask questions where necessary.

Task 5: For more clarification, read the **key readings 6.5.1**

Key readings 6.5.1: Description of VoIP system upgrading.

- **Upgrading VoIP System (Hardware and Software)**

Upgrading a VoIP system involves replacing outdated hardware components or upgrading software to improve performance, add features, or enhance security. It's a strategic decision that requires careful planning and execution.

- **Hardware Upgrading**

  ✓ **Involves:** Replacing physical components like IP phones, gateways, or servers.
  ✓ **Benefits:** Improved performance, enhanced features (e.g., HD audio, video conferencing), increased scalability, and longer lifespan.
  ✓ **Challenges:** Higher upfront costs, potential downtime, and compatibility issues with existing software.

- **Software Upgrading**

  ✓ **Involves:** Updating the IP PBX software or other VoIP applications.
  ✓ **Benefits:** Lower costs, faster implementation, access to new features (e.g., unified communications), and security patches.
  ✓ **Challenges:** Compatibility issues with older hardware, potential data migration challenges, and user training requirements.

- **Key Considerations for Upgrading**
    - ✓ **System assessment:** Evaluate the current system's performance, capacity, and limitations.
    - ✓ **Cost-benefit analysis:** Weigh the costs and benefits of hardware and software upgrades.
    - ✓ **Scalability:** Consider future growth and expansion needs.
    - ✓ **Compatibility:** Ensure compatibility between new and existing components.
    - ✓ **User impact:** Minimize disruptions and provide necessary training.
    - ✓ **Phased approach:** Consider upgrading in stages to reduce risks.

Points to Remember

Upgrading a VoIP system involves replacing outdated hardware components or upgrading software to improve performance, add features, or enhance security. It's a strategic decision that requires careful planning and execution.

Application of learning 6.5.

XYW Company has been using a VoIP system for several years, and it's now experiencing performance issues, limited features, and security vulnerabilities. To address these challenges and improve overall communication, you are tasked with planning and executing a VoIP system upgrade.

Tasks:

1. Assess the current VoIP system's performance, limitations, and future needs.
2. Develop a comprehensive upgrade plan, including hardware and software components.
3. Evaluate potential vendors and solutions.
4. Implement the upgrade process with minimal disruption to business operations.
5. Test the upgraded system to ensure optimal performance and functionality.
6. Develop a post-upgrade support and maintenance plan.

Learning outcome 6 end assessment

Theoretical assessment

Q1. Choose correct answer.

i. What is the most appropriate backup frequency for critical VoIP system components like the PBX server?

a) Daily
b) Weekly
c) Monthly
d) Annually

ii. A full backup is the most efficient backup method in terms of storage space and time.

a) True
b) False

iii. Which of the following is NOT a common VoIP hardware component?

a) VoIP phone
b) Router
c) Printer
d) PBX

iv. Firewalls are essential for protecting VoIP systems from external threats.

a) True
b) False

v. What is the primary purpose of encryption in VoIP communications?

a) Improving call quality
b) Reducing network congestion
c) Protecting call data fSrom eavesdropping
d) Increasing call setup speed

vi. Which of the following is the first step in VoIP corrective maintenance?

a) Implementing solutions
b) Identifying the root cause
c) Restoring normal system operation
d) Documenting corrective actions

vii. User feedback is essential for effective VoIP corrective maintenance.

a) True

b) False

viii. What is the primary benefit of upgrading a VoIP system?

a) Increased hardware costs

b) Reduced system performance

c) Improved features and functionality

d) Increased security vulnerabilities

ix. A migration plan is crucial for a smooth VoIP system upgrade.

a) True

b) False

Q2. Match the VoIP software component with its function.

SN	VOIP Software component	Functions
1	a) PBX	i. Manages call routing and features
2	b) Softphone	ii. Software-based phone application
3	c) Call manager	iii. Handles incoming calls and directs them to appropriate extensions
4	d) Auto-attendant	iv. Central switching system for managing voice calls

Practical assessment

AAAA Company has been using a VoIP system for several years to support daily operations. To ensure optimal performance, reliability, and security, you are responsible for maintaining the VoIP system.

Tasks:

1. Develop and implement a proactive maintenance plan.
2. Monitor system performance and identify potential issues.
3. Troubleshoot and resolve VoIP system problems.
4. Manage user accounts and phone configurations.
5. Maintain system security through regular updates and patches.
6. Create and maintain system documentation.

END

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