Title: Study on EPABX & SONET SDH.

Problem Statement:

1. Experiment on core networking voice and data transport over SONET/ SDH Network
2. Experiment on EPABX(IP-PBX), ISDN and ADSL system for dial -on-Demand Network
3. Demonstration on voice over IP and IP Telephony
4. Demonstration and performance measurement of dual frequency over a single channel using DSLAM and Splitter.

Components Required:

1. Cisco SDH ADMs (ONS 15305)
2. Cisco ATM Switches MGX 8830/B)
3. Cisco Routers (2881)
4. Cisco L2 switches
5. Optical Fiber
6. Fiber management trays
7. Desktop pc's

Theory & Background:

Experiment on core networking voice and data transport over SONET/ SDH Network

SDH (Synchronous Digital Hierarchy): Synchronous Digital Hierarchy (SDH) is a standardized telecommunications technology used for transporting digital signals over optical fiber networks. It is an international standard developed to replace the previous asynchronous transmission methods.

SDH provides a flexible and efficient way to transport large volumes of data, including voice, video, and data traffic, at high speeds over long distances. It achieves synchronization by employing a synchronous timing reference known as a synchronous payload envelope (SPE). Features of SDH:

Synchronization: SDH ensures precise synchronization of data transmission across the network, allowing for accurate multiplexing and demultiplexing of signals.
Multiplexing: SDH supports multiplexing of different data streams, allowing multiple signals to be combined into a single high-speed transmission stream.
Protection and Restoration: SDH includes built-in features for network protection and restoration. It can automatically reroute traffic in the event of a network failure, helping to ensure high reliability and availability.
Flexible Bandwidth Allocation: SDH allows for flexible allocation of bandwidth, enabling network operators to efficiently manage network resources based on changing traffic demands

ADM (Add drop multiplexer): An Add-Drop Multiplexer is a key component in synchronous optical networks, such as Synchronous Digital Hierarchy (SDH) or Synchronous Optical Networking (SONET), and it plays a crucial role in managing the flow of data within these networks. Some features

Addition: One of the primary functions of an ADM is to add (or inject) new data signals into the existing optical network. This addition can happen at specific points along the network where new data needs to be introduced without disturbing the ongoing traffic.
Dropping: Similarly, an ADM can selectively drop (or extract) specific data signals from the optical network. This capability allows for the removal of specific data streams at certain points in the network without affecting the rest of the traffic.
Multiplexing: An ADM combines multiple incoming optical signals onto a single fiber optic link for transmission. These signals may come from various sources, such as different geographical locations or different services within the network.

Procedure:

- At first establish the connection between two setups of LANs through optical fiber optic ring & observed their Ip Address.
- Then configured the SDH-ADMs using Cisco edge craft software.
- Then open VLAN setting in that software and map one LAN and one to the VLAN.
- Then set the BW of the port mapped to WAN as 2 channels of 4.34 Mbps each.
- Then open WAN to SDH mapping in that software & map the WAN port to the corresponding SDH channel for both upstream & downstream.
- Now SDH in configured and checked by those following commands
- C: \ > cd/
- C: \ > ping 10.6.1.5
- C: \ > ftp 10.6.1.5
- C: \ > user, password
- C: \ > bin, hash, nids.pdf
- Now, note the throughputs at both source and the destination and compare them.

Results:
1. At the source PC, we get the throughput as 516.16 Kbytes/sec.
2. At the destination PC, we get the throughput as 515.07Kbytes/sec.

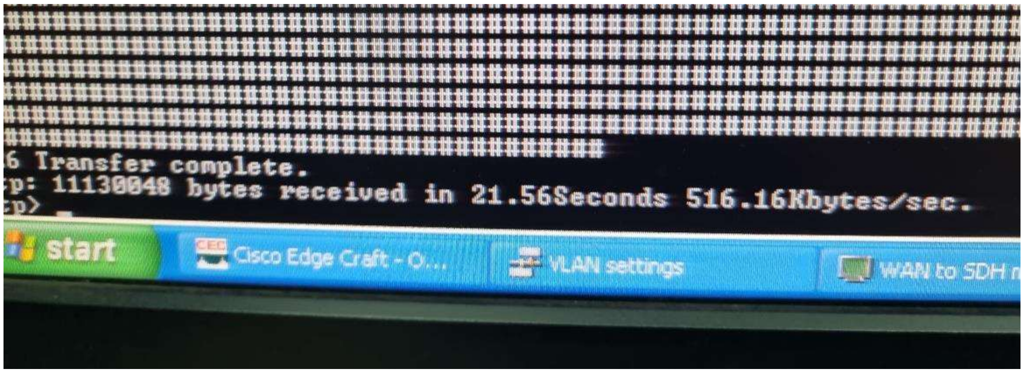

Fig: Source Pc throughput.

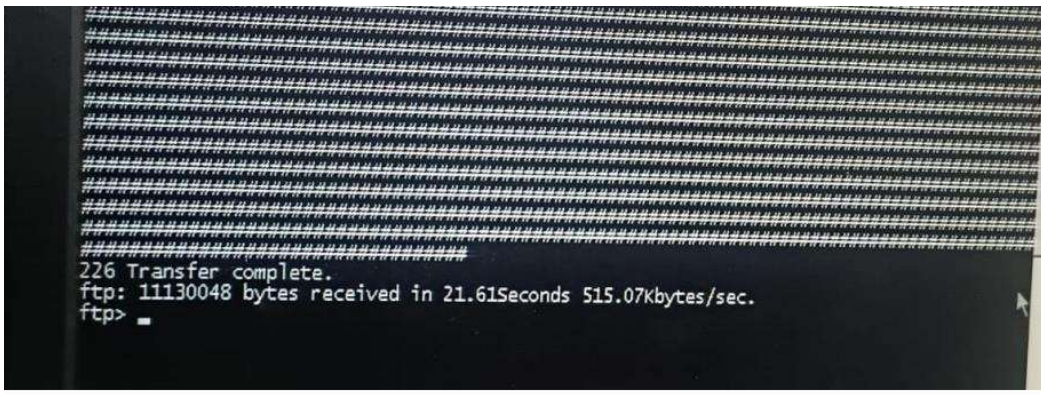

Fig: Destination Pc throughput.

Experiment on EPABX(IP-PBX), ISDN and ADSL system for dial-on-demand Network.

To conduct an experiment on EPABX (IP-PBX), ISDN, and ADSL systems for a dial-on-demand network, we need to set up a testbed environment that simulates the functionalities and interactions of these components.

Equipment needed:

1. EPABX/IP-PBX system with IP telephony capabilities.
2. ISDN terminal adapters and ISDN-compatible devices.
3. ADSL modems and routers.
4. Analog/digital phones.
5. Test phone lines or simulated telephone connections
6. Network cables and connectors

EPABX/IP-BPX: EPABX (Electronic Private Automatic Branch Exchange) and IP-PBX (Internet Protocol Private Branch Exchange) are both telephone systems used within organizations to manage internal and external phone calls. While EPABX relies on traditional circuit-switched telephony, IP-PBX leverages Internet Protocol (IP) networks for voice communication.

- Traditionally, PBX systems were manual switchboards operated by human operators. EPABX systems evolved from PBX systems, incorporating electronic automation for call routing and management.
- EPABX systems automate call routing, eliminating the need for human operators to manually connect calls.
- EPABX systems facilitate internal communication within an organization. Users can make calls to colleagues within the same office or branch without dialing external phone numbers.
- EPABX systems offer various call handling features, including call forwarding, call transfer, conference calling, all waiting, voicemail, and automated attendant (IVR) systems.
- EPABX systems include security features to protect against unauthorized access and misuse of phone lines. Encryption, authentication, and access control mechanisms.

An **IP-PBX** (Internet Protocol Private Branch Exchange) is a private branch exchange (PBX) telephone system that operates over IP networks, such as a local area network (LAN) or the internet. It leverages VoIP (Voice over Internet Protocol) technology to transmit voice data packets over the network, enabling advanced communication features and integration with other IP-based services.

- advanced communication features and integration with other IP-based services. This allows for more efficient use of network infrastructure and enables a wide range of advanced features.
- Unlike traditional PBX systems that rely on hardware components, IP PBX systems are often software-based.
- IP PBX systems can help organizations reduce communication costs, especially for long-distance and international calls, by routing them over IP networks instead of traditional phone lines
- With IP PBX systems, employees can access their office phone system from anywhere with an internet connection, allowing for remote work flexibility.
- Security is a critical consideration for IP PBX systems. Implementing encryption, access control, and other security measures can help protect against unauthorized access, eavesdropping, and other threats to communication privacy and integrity.

ADSL (Asymmetric Digital Subscriber Line) is a type of digital communication technology that enables high-speed data transmission over existing copper telephone lines. It is widely used for broadband internet access in residential and small business environments.

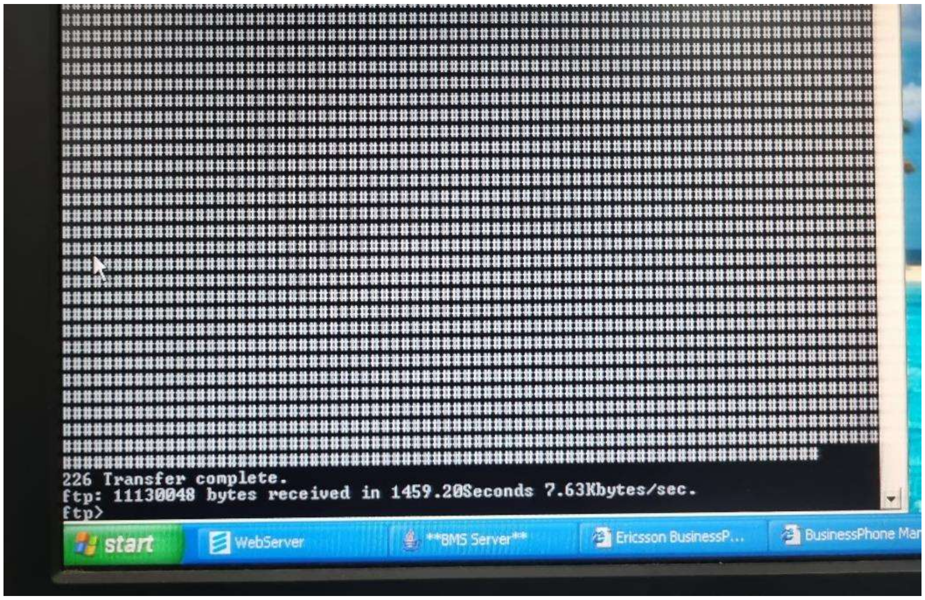

Fig: Data rate on ISDN.

Experiment on - Demonstration on voice call over IP and IP telephony

To demonstrate voice calls over IP (VoIP) and IP telephony, we set up a simple test environment using VoIP software and IP telephony devices.

Component Required:

- Ericsson EPABX system.
- Cisco IP phone.
- Cisco Routers.
- Call manager administration.

Voice over internet protocol (VoIP):

It is a technology that allows users to make voice calls over an IP network, such as the internet, instead of using traditional telephone lines. VoIP converts analog voice signals into digital data packets and transmits them over the internet in real-time.
Packet-Switched Technology: VoIP utilizes packet-switched technology to transmit voice data over IP networks. Voice signals are digitized, compressed, and broken down into small data packets for transmission.

Codec Compression: VoIP uses audio codecs (coder-decoder) to compress voice data and optimize bandwidth usage. Codecs such as G.711, G.729.

Real-Time Communication: VoIP enables real-time communication, allowing users to make instant voice calls, conduct video conferences, and send multimedia messages over the internet.
Integration with Other Services: VoIP can integrate with other communication services and applications.
Cost-Effective: VoIP typically offers lower call rates compared to traditional telephone services, especially for long-distance and international calls.
Flexibility and Mobility: VoIP enables users to make calls from any location with an internet connection, making it suitable for remote and mobile workers.

Hyper Terminal:

HyperTerminal is a terminal emulation program that allows users to communicate with other devices over various communication ports such as serial, COM, or TCP/IP. It was originally developed by Hilgraeve Inc. and was included with various versions of the Windows operating system up to Windows XP. HyperTerminal provides a simple interface for configuring and managing connections to serial devices, network devices, and other communication equipment.

Terminal Emulation: HyperTerminal emulates various terminal types, including ANSI, VT100, VT220, and others, allowing users to interact with remote systems or devices using terminal commands.

TCP/IP Connectivity: HyperTerminal can establish TCP/IP connections to network devices using Telnet or direct TCP connections.

File Transfer: HyperTerminal includes basic file transfer capabilities for sending and receiving files over serial or TCP/IP connections. It supports protocols such as Xmodem, Ymodem, and Zmodem for file transfer.

Serial Communication: HyperTerminal enables communication with devices connected to serial ports (COM ports) on the computer.

Scripting: HyperTerminal supports basic scripting functionality using Visual Basic Scripting Edition (VBScript), allowing users to automate tasks or perform custom actions during communication sessions.

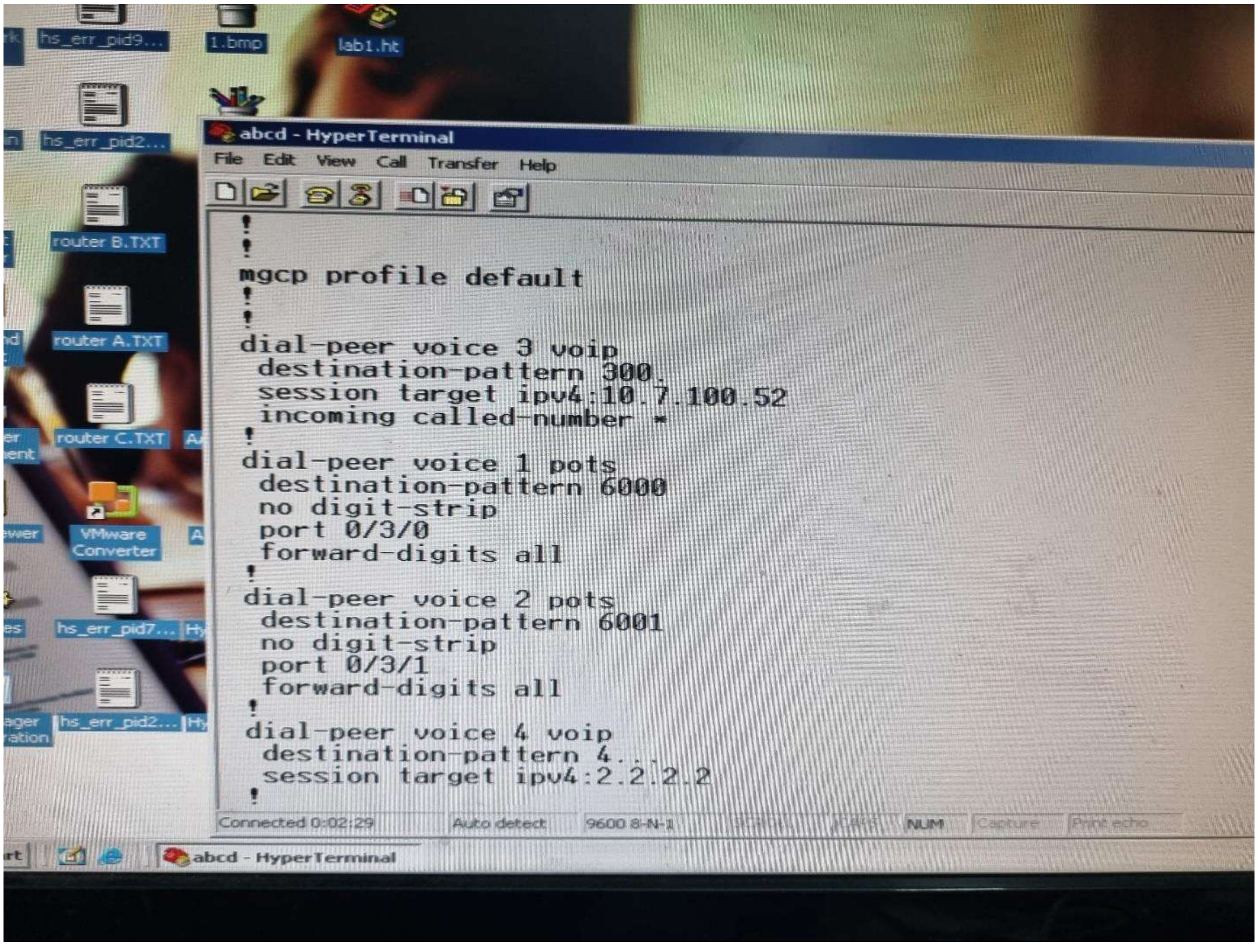

Fig: Hyper terminal showing devices connected within the network.

Call Manager Administration:

Call Manager Administration, often referred to as Cisco Unified Communications Manager Administration or simply CUCM Administration, is a web-based management interface used to configure, manage, and monitor Cisco Unified Communications Manager (CUCM) systems.

CUCM is a key component of Cisco's Unified Communications suite, providing call processing, voice messaging, mobility, and conferencing services for organizations of all sizes.

PROCEDURE:

- Firstly, establish the network interconnection as shown in the fig. and note the IP addresses of each required port and phone number of each phone in the network test bed.
- Then open the cisco call manager and study the IP-telephony management schemes used.

- Make calls from different IP-phones to test the communication.

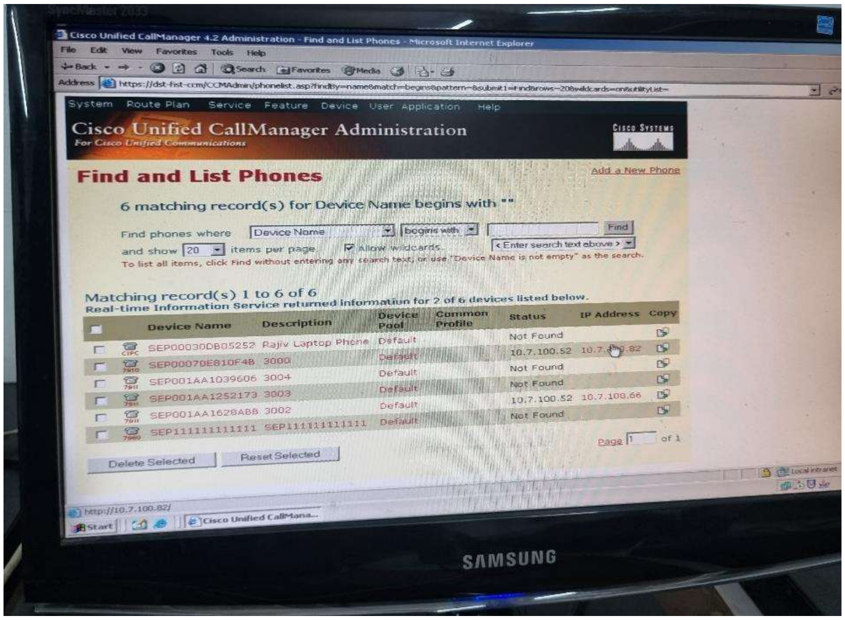

Fig: Cisco call manager system.

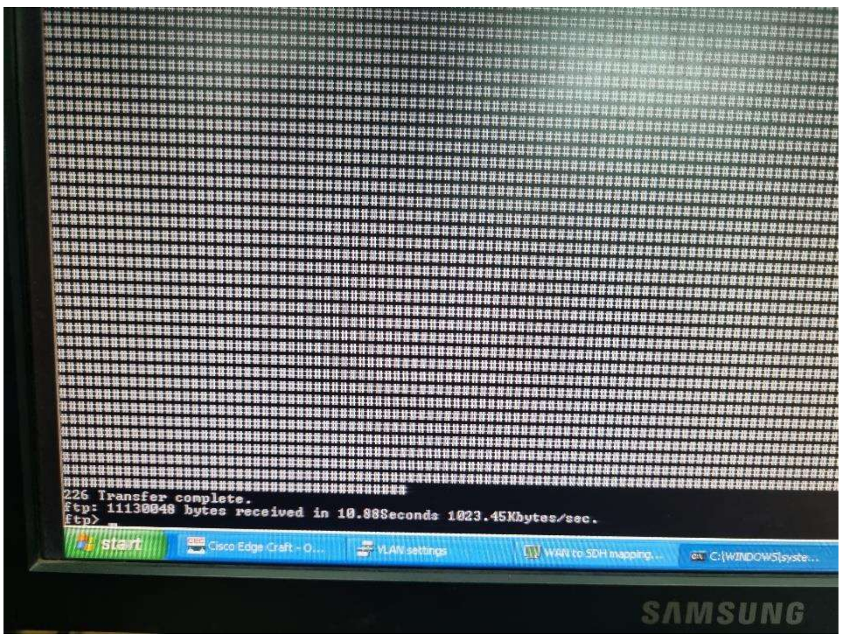

Fig: Date rate on voip.

We checked how the IP- phones are connected in the network by checking the IP address on the Cisco Call
Manager and made calls from different IP-phones to test the communication.
Data Transfer: We get the data rate as 1023.45 Kbytes/sec.

Experiment- Demonstration and performance measurement of dual frequency over a single channel using DSLAM-splitter.

COMPONENTS REQUIRED:

1. Ericsson EPABX system
2. Cisco IP phones
3. Cisco Routers
4. D-Link DSLAM

A DSLAM (Digital Subscriber Line Access Multiplexer) splitter, also known simply as a splitter or microfilter, is a device used in DSL (Digital Subscriber Line) broadband internet connections to separate voice and data signals on a telephone line.

RESULT:

We get the data rate as 92.99 Kbytes/sec in 119.69sec.

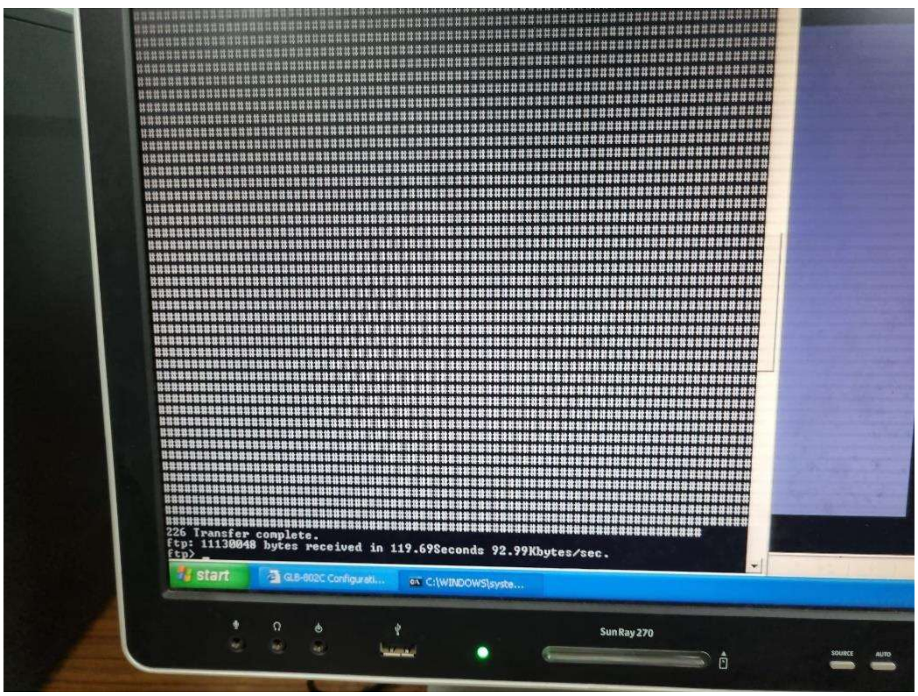

Fig: Data rate in DSLAM splitter system.