VOICE
SOLUTIONS

NGN

Telephone-Handbook

NEXT GENERATION NETWORK

Table of Contents

1. Introduction

The emergence of Next Generation Network (NGN) solutions has had a significant impact on the way businesses deploy telephony services. Recent advances in networking technology has stepped up the industry's shift from PSTN towards Voice over IP (VoIP) based services.

<div style="background-color:orange">

NGN Technology: A step forward for corporate sector

</div>

Nayatel (Pvt) Ltd. being a facilities based carrier provides converged Triple Play services (Voice, Video, and Data) on Pakistan's first Fiber-to-the-user (FTTU) network. Nayatel's Next-Generation-Network (NGN) is new, evolution-oriented; telecommunication network architecture based on packet transport and dedicated control and service layers. The objective of NGN is to have a single network for all telecommunication services to provide high efficiency and multi-services. It supports legacy applications of Plain Old Telephony System (POTS) and also new and enriched applications of future like real time video telephony, etc. In terms of quality of service, it provides top notch quality service, high quality voice, no dropped calls and enhanced voice features.

Nayatel NGN telephony hand book is particularly designed to give you insight into application scenarios of NGN telephony services in corporate environment. The information would hopefully be beneficial for customer organizations to access their corporate telephony needs and Nayatel's contribution to compliment value of their business operations.

Marketing Department
Nayatel (Pvt) Ltd.
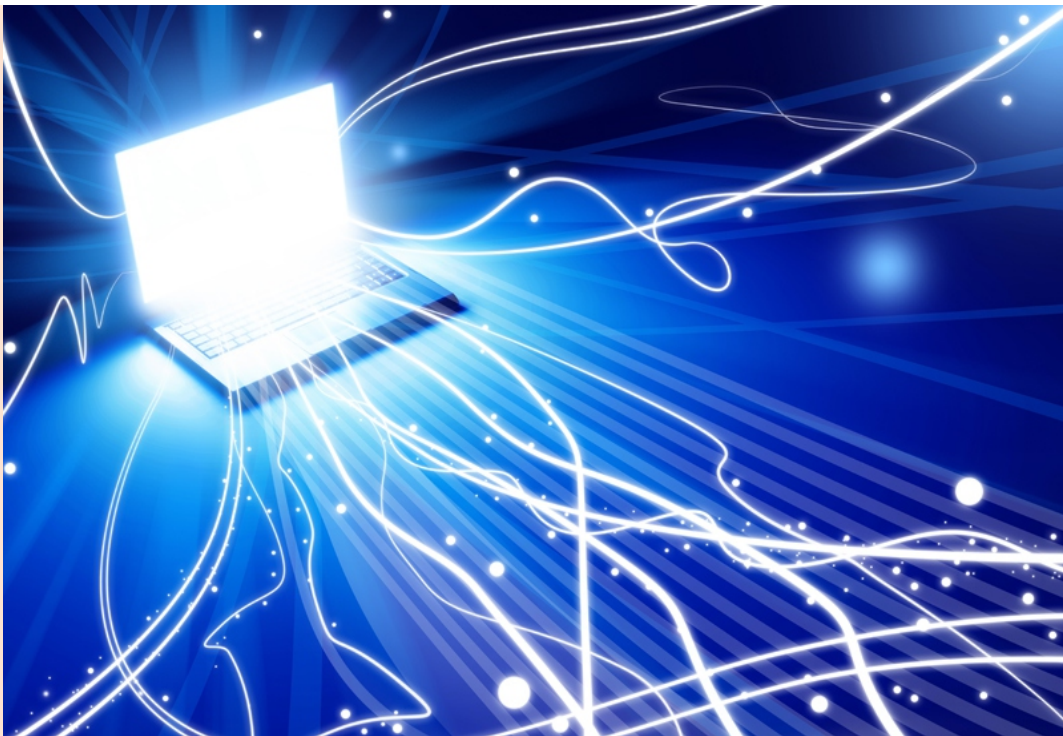
NEXT GENERATION NETWORK

2. Nayatel Hosted Exchange Solution

2a. Overview

Nayatel Hosted Exchange solution gives you the features and functionality of telephone systems without incurring huge cash outlays for traditional PBX equipment. Nayatel's Hosted Exchange, instead does that all virtually for you.

Nayatel HostEX Features

Corporate Office/Contact Center features

- Digital receptionist/Interactive Voice Response (IVR)
- Automatic Call Distribution (ACD)
- Call Queuing

- Ring Groups
- Music on Hold

Call Management Features

- Call Forwarding
- Call waiting indication
- Call transfer
- Call Monitoring
- Caller ID on extensions
- Call Parking

- Conference Calling
- Voice Mail
- Web based call statistics and reporting
- One exchange for multiple offices
- Direct Inward Dialing (DIDs)
- Call recording

Nayatel HostEX checklist

Requirements for Hosted Exchange Solution at customer premises

✓	Nayatel verified IP-Phones (Hard/soft phones)
✓	Managed Fast-Ethernet switch for the IP-Phone network
✓	Data and VoIP networks are physically or virtually (VLAN) separated. A separate virtual LAN is recommended for better voice quality
✓	UPS (power backup for Fast-Ethernet Switches, ATA & IP Phones)
✓	All Ethernet Cables are tagged on Fast-Ethernet Switch side
✓	All Ethernet Cables are tagged on IP-Phone Sides

2b. Hosted Exchange Architecture

PSTN

Nayatel NGN

Cell Phones
GSM/AIMS

Host EX

Virtual PABX | Virtual PABX

Nayatel
IP Network

ONT

Data and VoIP networks are physically or virtually (VLAN) separated

VOIP switch

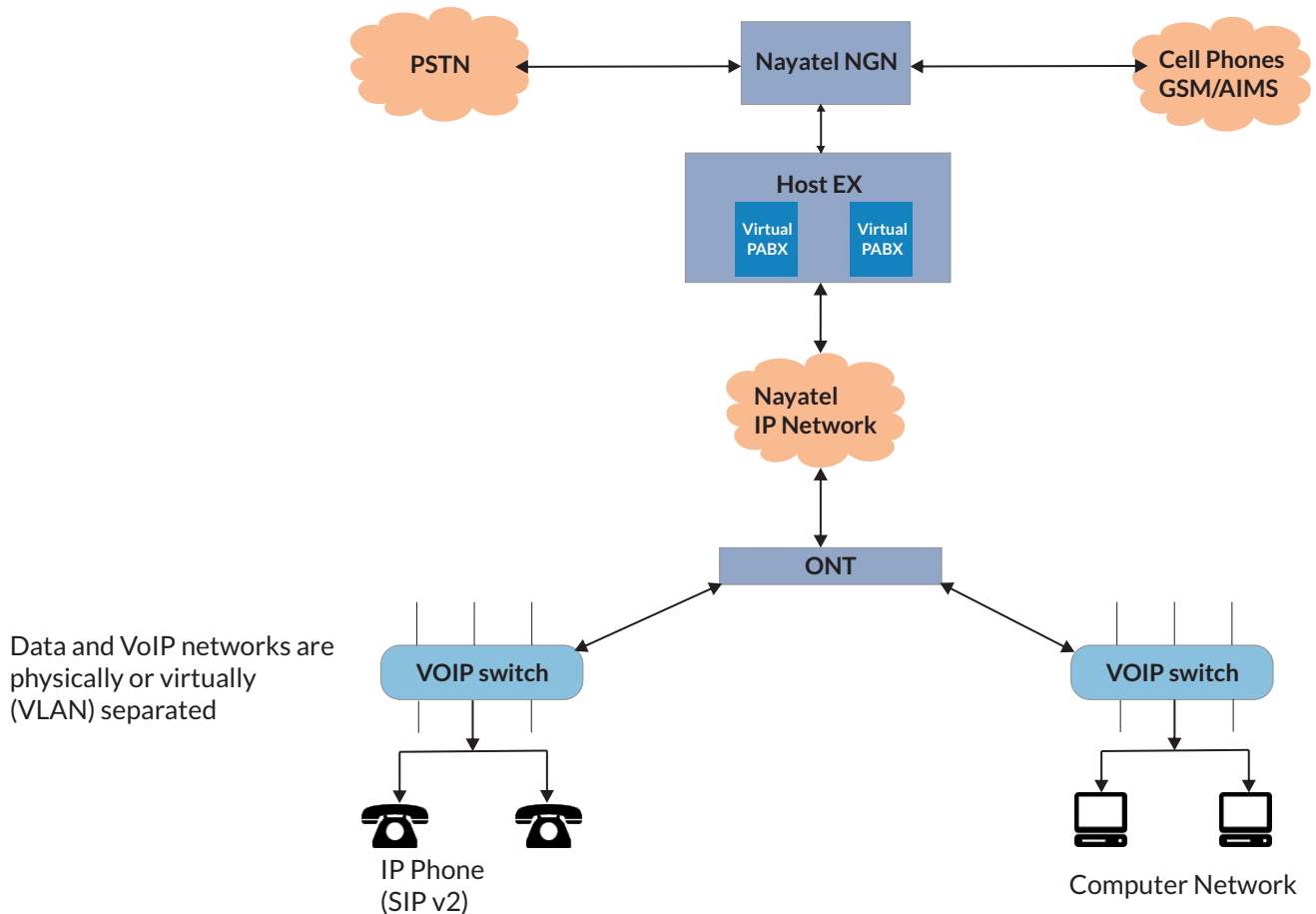VOIP switch

IP Phone
(SIP v2)

Computer Network

Figure 2.1

How Nayatel's HostEX (Hosted Exchange) works?

- Calls from PSTN/Cellular Networks terminate at Nayatel NGN switch
- Calls are then forwarded to customer's virtual PBX (located at Nayatel Data Center)
- Customer's Virtual PBX provides auto attendant features, DIDs, and conference calling etc
- Virtual PBX further routes incoming calls at desired extension numbers/IP phones (at Customer Premises)
- "Follow me" feature is available if the call is unanswered or Manager/Administrator wants incoming calls to be attended at any other premises/remote location
- All outgoing calls from customer premises are received at NGN switch, and are further routed to interconnect networks

2c. Hosted Exchange Application Scenarios

Nayatel HostEX ⟷ **Nayatel IP Network**

ONT

Ethernet Switch

Analog Telephone Adapter

RJ11   RJ11

Figure 2.3 a

a. Analog Telephony Adapter (ATA)

- ATA interface with the switch allows users to connect conventional analogue phone sets instead of using IP phones

- ATA is a device used to connect one or more standard analog telephones to a Voice over IP based network. An ATA usually takes the form of a small box with a power adapter, one Ethernet port.
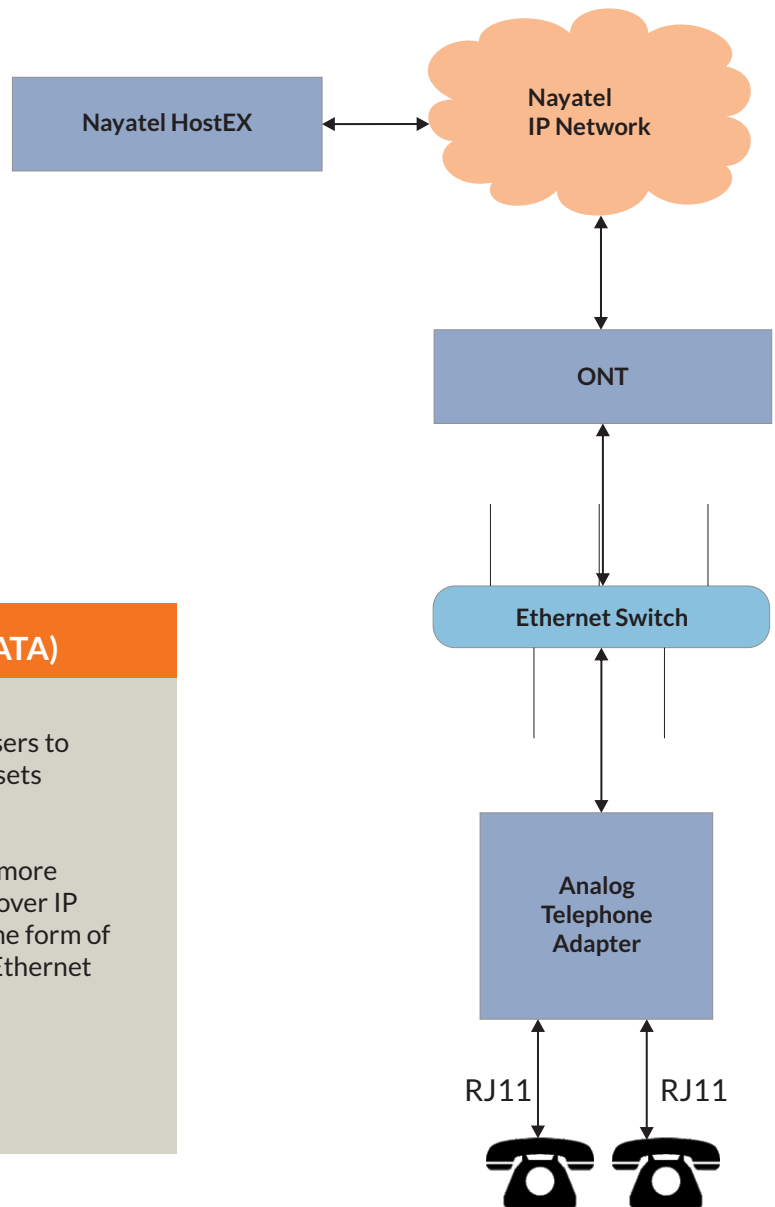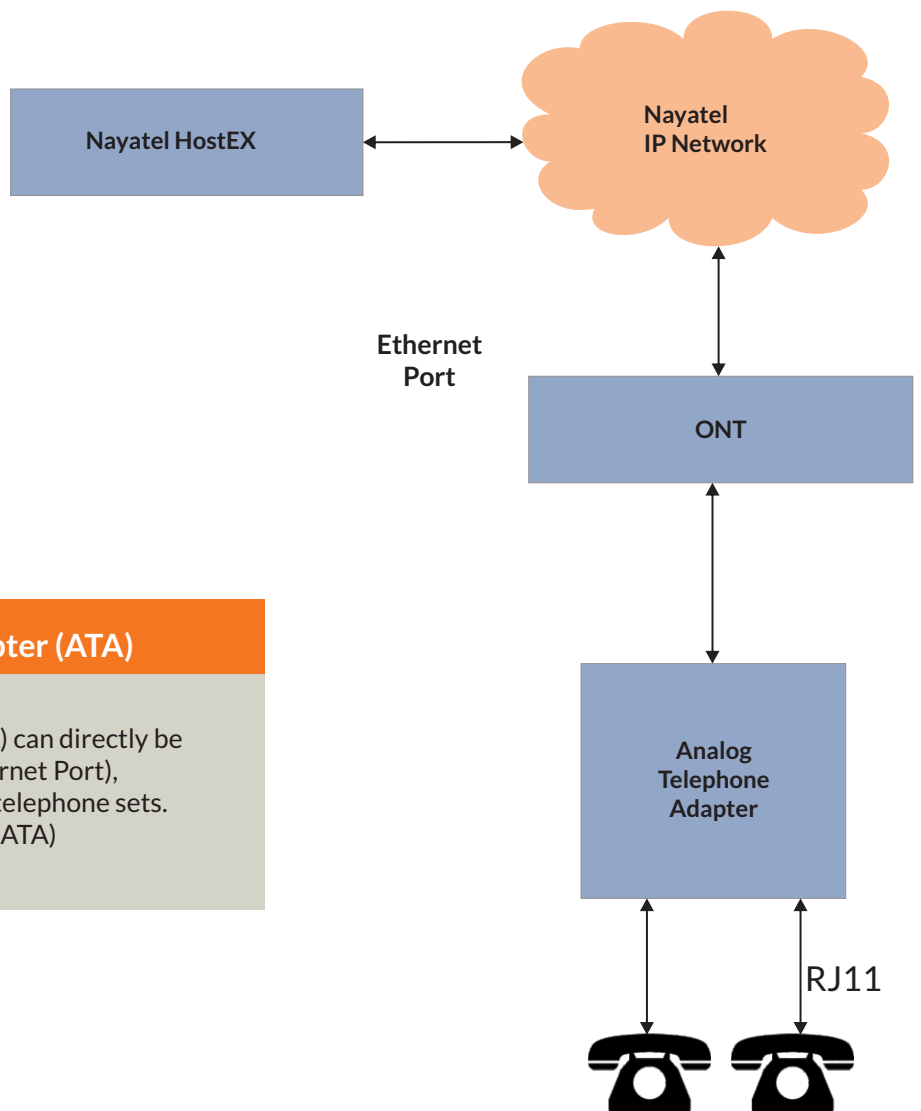
2c. Host Exchange Application Scenarios

Nayatel HostEX

Nayatel IP Network

Ethernet Port

ONT

b. Analog Telephone Adapter (ATA)

- Analog Telephony Adapter (ATA) can directly be connected with your ONT (Ethernet Port), allowing you to connect analog telephone sets. (Equivalent to ports available in ATA)

Analog Telephone Adapter

RJ11

Figure 2.3 b
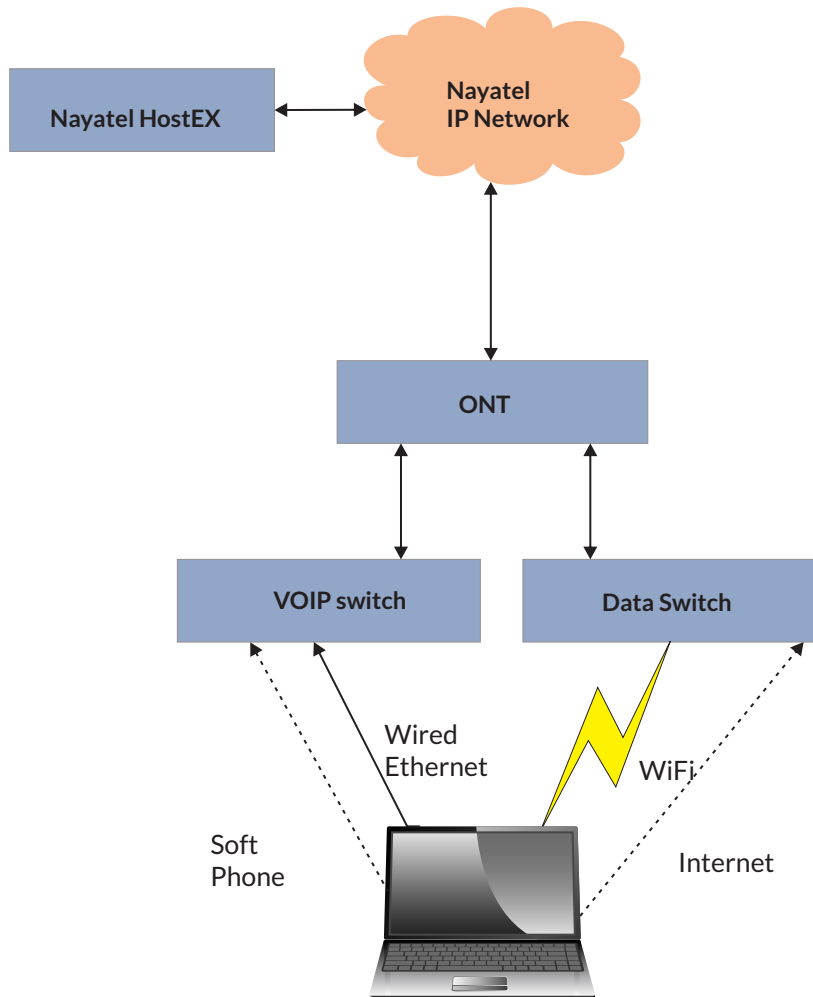
2c. Host Exchange Application Scenarios

Figure 2.4

c. Internet & voice integration 1

- In above mentioned scenario contact center employees can use their PC for data and voice applications simultaneously.
- Customers can use soft phones (Nayatel verified) to make and receive calls via their desktop using wired internet, and internet can be used via WiFi network
- Data and VoIP networks are physically or virtually (VLAN) separated. A separate virtual LAN is recommended for better voice quality

2c. Host Exchange Application Scenarios

d. Internet & Voice Integration 2

- Under this scenario, employees' PCs can directly be connected with IP phone's Ethernet Port
- To execute this scenario, IP phone must support VLAN tagging capability (voice/data packets are tagged with different VLAN IDs to segregate the traffic)
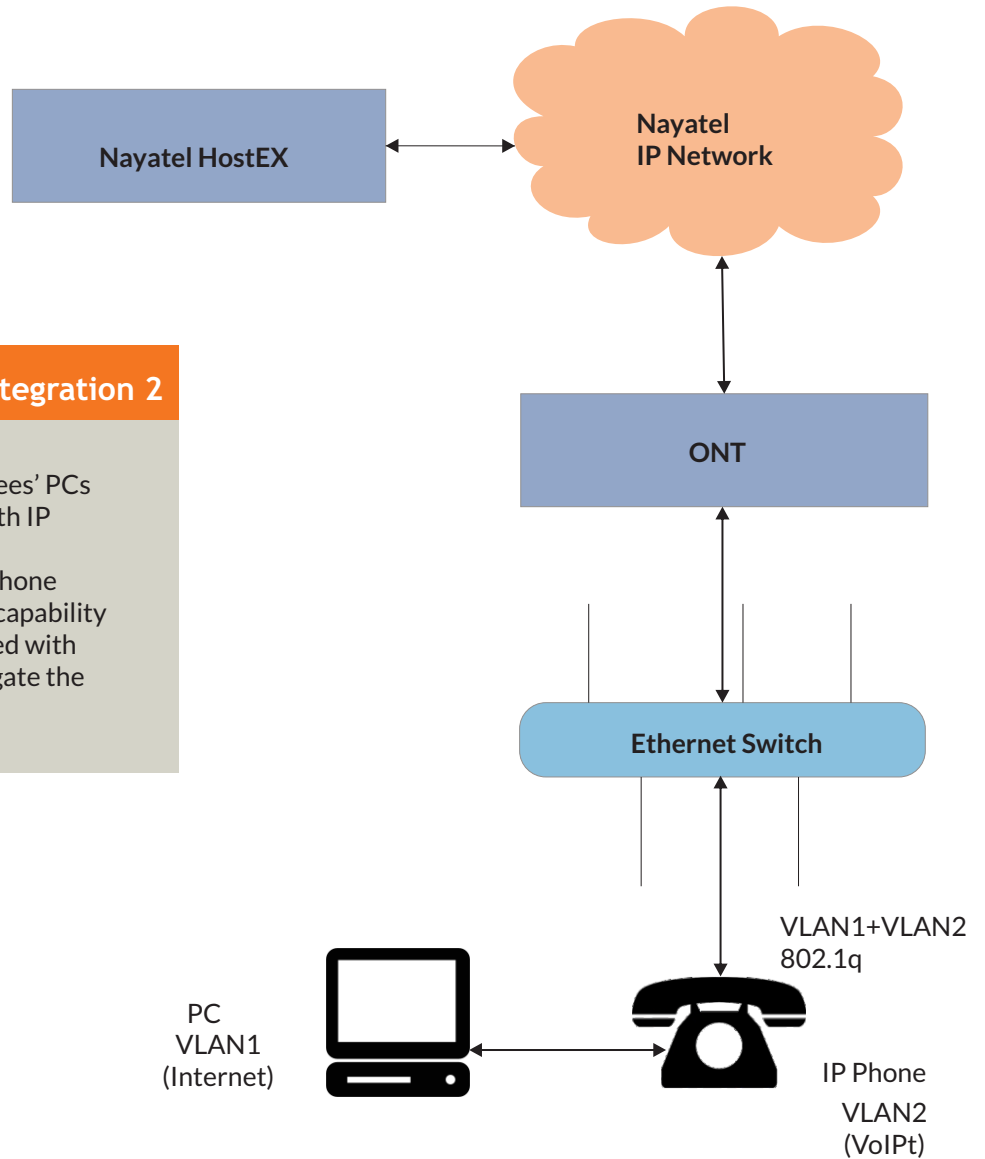
Nayatel HostEX

Nayatel
IP Network

ONT

Ethernet Switch

VLAN1+VLAN2
802.1q

PC
VLAN1
(Internet)

IP Phone

VLAN2
(VoIPt)

Figure 2.5
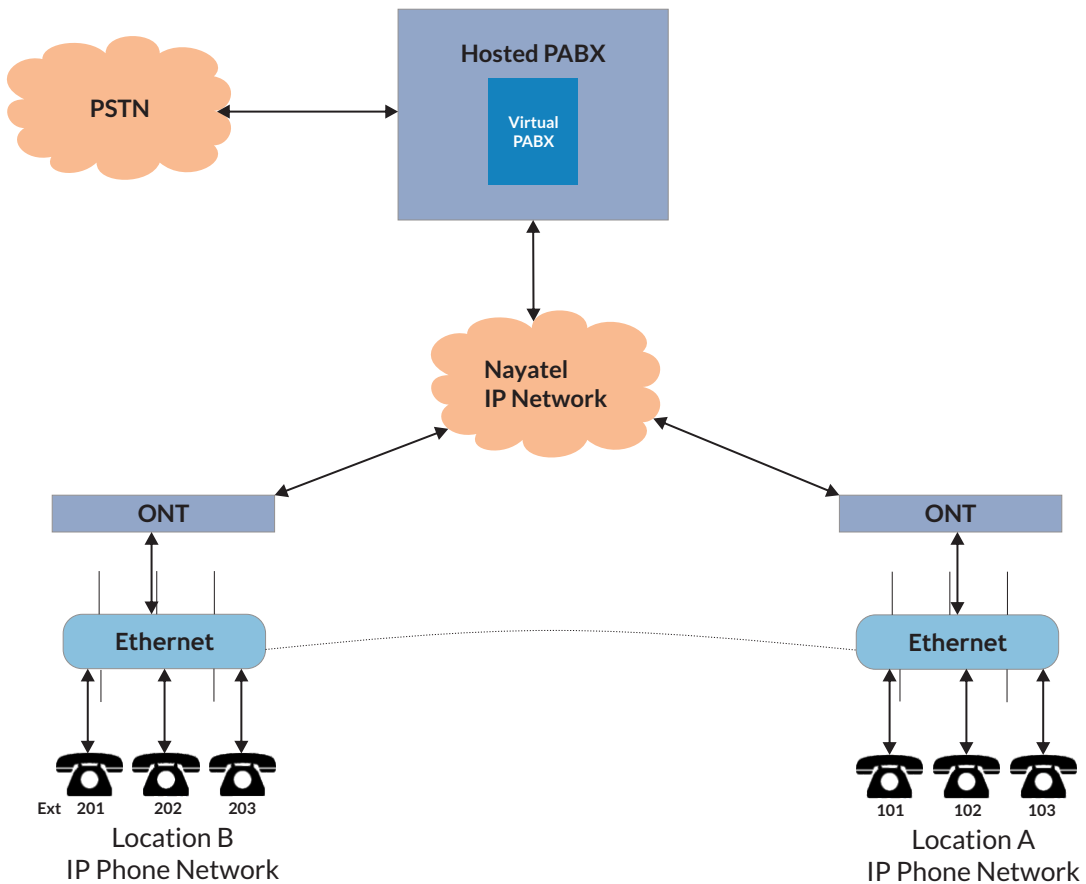
2c. Host Exchange Application Scenarios

Figure 2.6

e. Interoffice communication (without local call charges)

- Fiber connectivity is required at both ends (P2P link)
- The feature allows the users (Location A&B) to commute with extensions/IP phones at both ends/ locations without bearing local call charges
- Nayatel virtual PBX (Nayatel Data Center) is shared between both locations

3. SIP (Session Initiation Protocol)
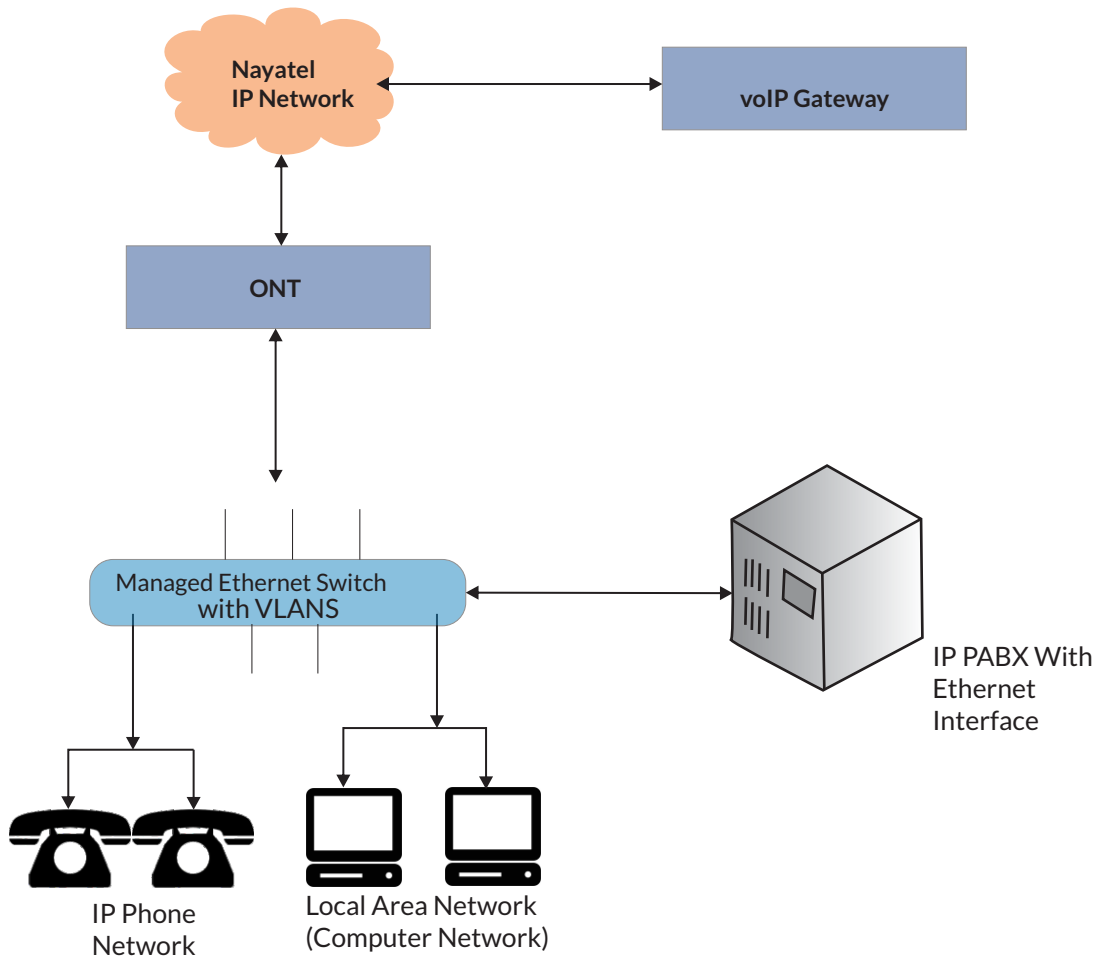
a. SIP Architecture

Figure 3.1

a. SIP (Session Initiation Protocol)

- The Session Initiation Protocol (SIP) is an application-layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants. It can be used to create two-party, multiparty, or multicast sessions that include Internet telephone calls, multimedia distribution, and multimedia conferences.
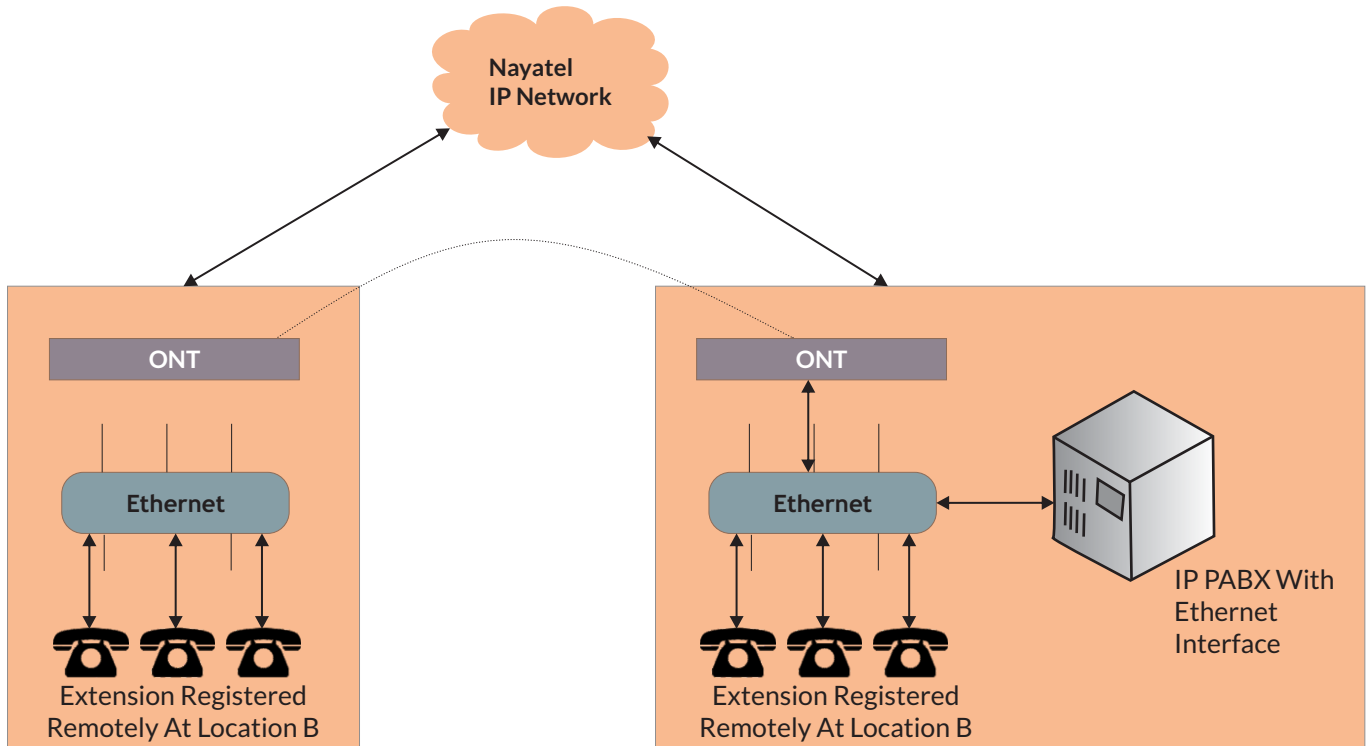
a. How SIP works?

- Calls from PSTN/Cellular networks terminate at Nayatel NGN Switch (VoIP gateway), which are forwarded to IP PBX (situated at customer premises)
- Customer can either use SIP enabled IP phones, Soft phone, or analogue phones (with ATA) to commute (which are registered/connected to IP PBX)
- All outgoing calls from customer's IP PBX through Nayatel's IP Network are received at NGN switch, and further routed to interconnect networks

b. SIP Interoffice Communication

Nayatel IP Network

ONT

Ethernet

Extension Registered Remotely At Location B

ONT

Ethernet

IP PABX With Ethernet Interface

Extension Registered Remotely At Location B

b. SIP Interoffice communication (without local call charges)

- Fiber connectivity is required at both ends (P2P link)
- The feature allows the users (Location A&B) to commute with extensions/IP phones at both ends/ location without bearing local call cost
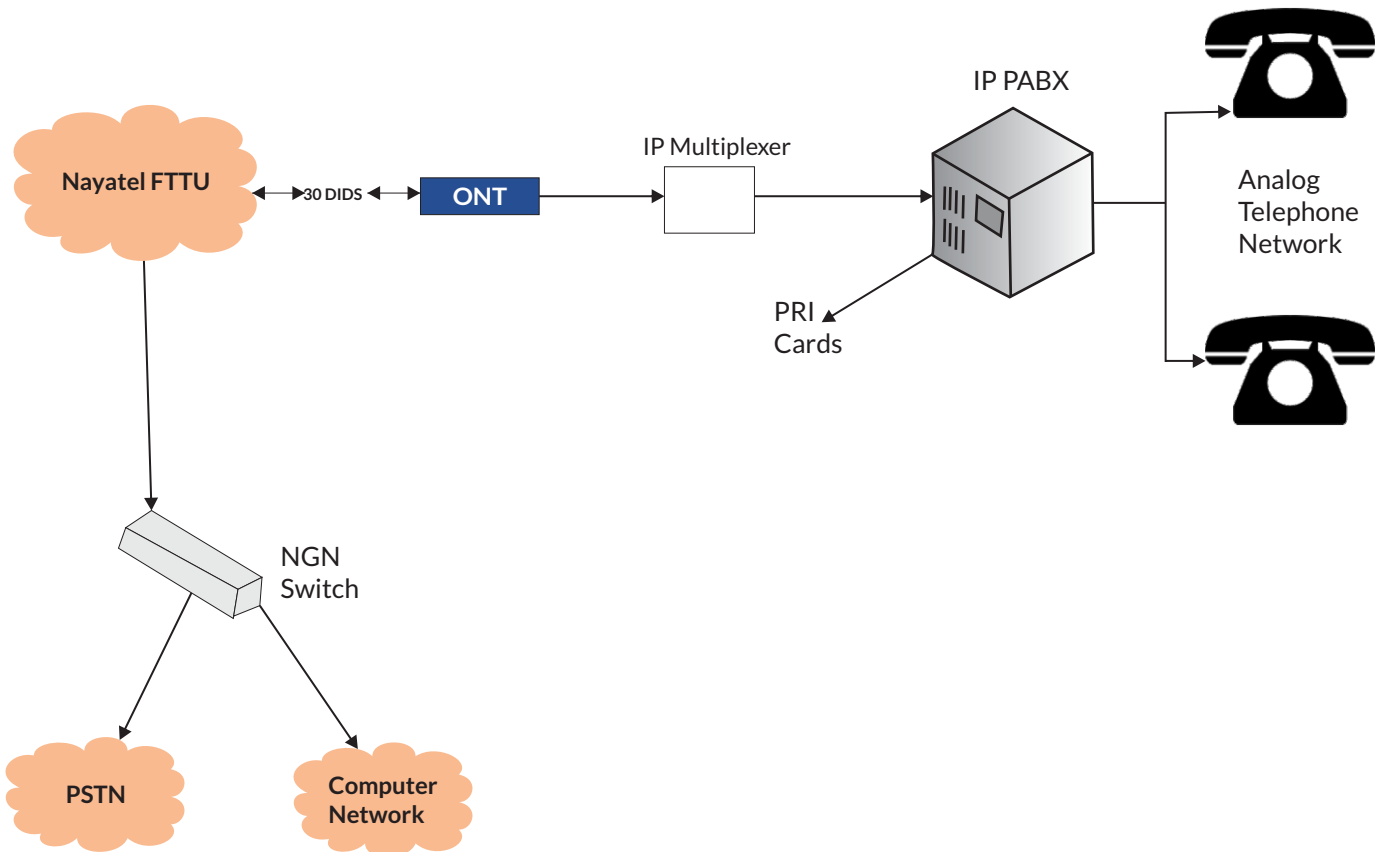
4. Primary Rate Interface(PRI)

a. PRI Architecture

Figure 4.1

a. PRI

- Calls from PSTN/Cellular networks terminate at Nayatel NGN Switch
- Calls are then forwarded at PBX (situated at customer premises)
- IP Mux is used as an interface between ONT (optical Network Terminal) and PBX, to convert circuit switching into VoIP and vice versa
- PRI provides 30 dedicated voice channels over a single physical link
- All outgoing calls from customer's PBX through Nayatel's IP Network are received at NGN switch, and further routed to interconnect networks
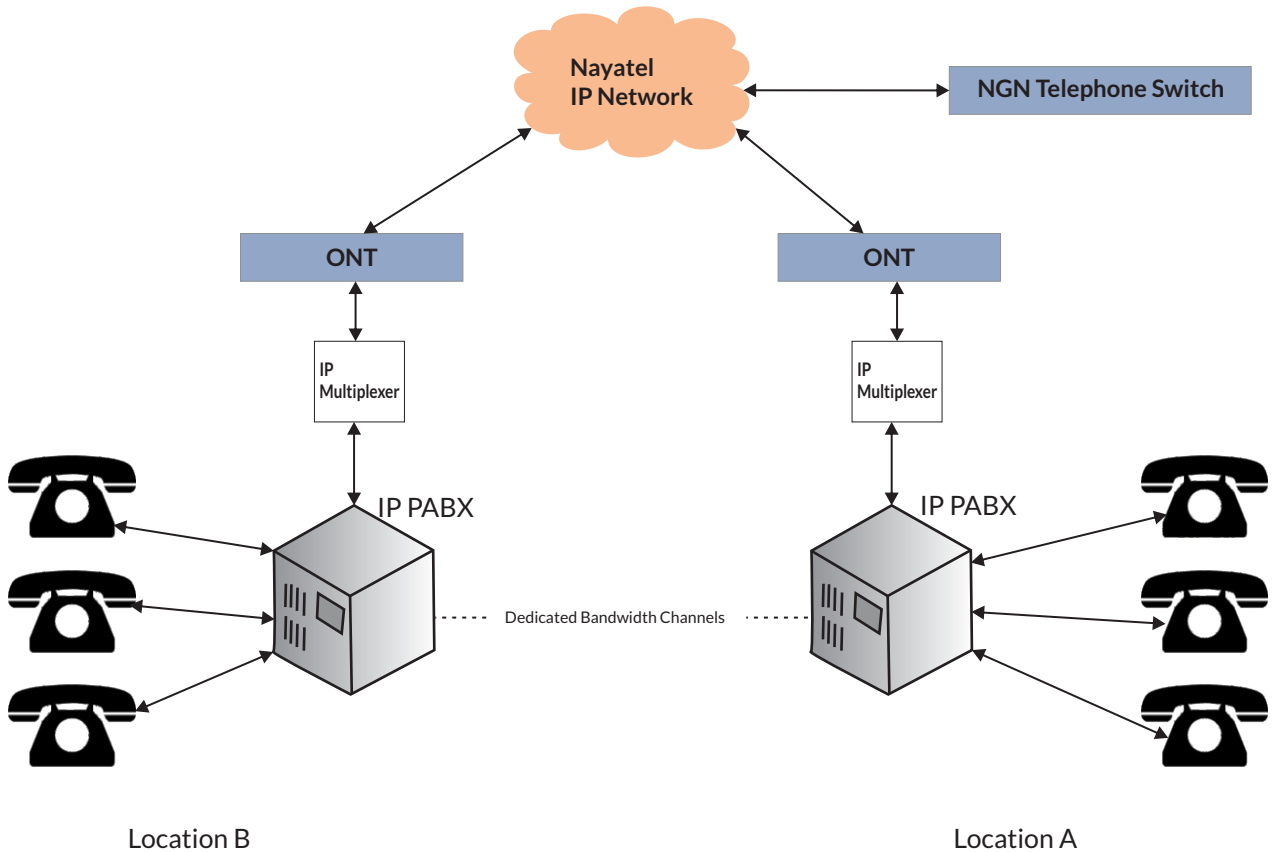
a. PRI Interoffice Communication

Figure 4.2

d. PRI Interoffice Connectivity

- Fiber connectivity is required at both ends (P2P link)
- The feature allows the users (Location A&B) to commute with extensions/IP phones at both ends/locations without bearing local call charges.

5. Customer Services

In case you need further information on Nayatel's NGN telephony solutions and their adaptability in your organization, or if your would like our customer service representative to guide you of a particular service feature or to assess your needs, please contact by either of the following means;

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Sales:.....................................sales@nayatel.com
General Information:....... Info@nayatel.com
Technical Support:............support@nayatel.com

Website
http://www.nayatel.com
NGN telephony section on Nayatel's web portal provides complete overview on FTTU voice portfolio, service features, FAQs, and tariff etc.
In case you need further information and guidance on NGN Telephony services, you can also submit the online information request form.

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