Network Configuration and VoIP Load Balancing of Golden Telecom

By

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B.Sc. in Electronics and Telecommunications Engineering

Submitted to the Dept. of Electronics and Communications Engineering in Partial Fulfillment of the Requirements for the Degree of

Master of Science in Telecommunication Engineering

at

East West University Bangladesh

2014

DECLARATION

I hereby declare that this thesis is my own work and it has been written by me in its entirety. I have acknowledged all the sources of information which have been used in the thesis. This thesis, either in whole nor in part, has been previously submitted for any degree.

Signature of Supervisor

Signature of Author

Approval

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ABSTRACT

Internet is the groundbreaking discovery of 20th century. In this epoch of modernization we have observed massive development of the internet as well as the different programs supported by it. The aspect of VoIP comes at a stage when Telecommunications structure has hitherto turned into a complicated system accompanied by numerous challenging frameworks. VoIP significantly increases the ingrained difficulty by bearing a technology that allows various frameworks for conveying the identical voice and data service while remaining cynic to the key infrastructure.

The Internet Service Provides are facing huge load to ensure the appropriate services to their customers. But the bandwidth is limited for the vast number of users of the internet. So the traffic needs to be engineered to utilize them resultantly. Multipath routing is a common term in Traffic Engineering which is used to balance the load in the company of different gateways.

The objective of this thesis is to configure the network to improve network reliability, to configure a Quintum Tenor Gateway and to implement load balancing for **Golden Telecom Pte. Ltd.** which will ensure no disconnection or outage. A network simulator is used to run the unit tests against the configuration to ensure operation ability of the configuration to deploy. This project would help to ensure the proper and simultaneous usage of internet by not wasting any bandwidth.

Thesis Supervisor: **Professor Dr. Mohamed Ruhul Amin**
Thesis Co Supervisor: **Dr. Md. Zahedul Hassan**
Title: **Network Configuration and VoIP Load Balancing of Golden Telecom**

To My Parents
for their love and encouragement...

Acknowledgements

I would like to thank the following Faculty members:

- Dr. Md. Zahedul Hassan
- Dr. Mohamed Ruhul Amin
- Dr. M. Mofazzal Hossain
- Dr. Gurudas Mandal
- Dr. Feroza Begum
- Mr. Mustafa Mahmud Hussain
- Mr. Sarwar Jahan
- Md. Asif Hossain

A special thanks to my EWU batchmates and everyone involved with Golden Telecom.

Table of Contents

Chapter 1

Introduction

In this era of modern technology everyone expects to work efficiently in an environment where communication is easy, fast and cost effective. That distinctly specifies that we were in need of a cheaper and user friendly technology which should be simply available everywhere.

Voice over Internet Protocol, VoIP is a tremendous technology in the field of Telecommunication. VoIP is also known by terms such as Internet Telephony, Computer Telephony and Windows Telephony. In the matter of price VoIP has a logical difference than the analog telephone technology. VoIP already has gone beyond the initial phase and now it is being used so enormously by conventional users. The network configuration of Golden Telecom Pte. Ltd. has been done in this thesis.

VoIP is becoming extensive in the Telecommunications field, inaugurating new purposes and trimming prices compared to the conventional outputs they replace. IT executives of VoIP carriers should dynamically prepare for outage or network failure emergencies.

Network Load Balancing is often mentioned as Dual-WAN routing or Multihoming. Load Balancing is referred to as the aptness to balance the traffic over two Wide Area Network links. This aptness stabilizes network sessions such as VPN, Web, streaming media, Emails, terminal services etc. across different connections. It scatters the volume of bandwidth used by every Local Area Network user.

Generally load balancing is used to deliver network redundancy. So if any outage occurs in a Wide Area Network link then the network can be accessed via other link. Redundancy plays an important role in concomitance with high-priority business applications like VoIP, VPN etc.

Chapter 2

Structure of Golden Telecom

2.1 Brief of Golden Telecom

Since 2000 usage of VoIP has expanded dramatically. In a few years VoIP has reached the top as a alternative to traditional telephone service from an unorthodox position.

Golden Telecom Pte. Ltd was incorporated in Singapore in 2009 as a Service Based Operator. Golden Telecom is a licensed telecommunication company. It has regional office in Bangladesh, Hong Kong and UAE. Golden Telecom focuses on global wholesale of voice termination with highly efficient officials to ensure 24/7 effective service to all of its valuable partners. Golden Telecom carries a high quality premium class voice termination service with competitive pricing model which is so helpful for establishing good relations with long distance carriers, calling card operators and many reputed service providers.

Golden Telecom has partnerships with more than 460 carriers all over the world and handling 150 millions voice minutes per month through its carrier grade Multi protocol VoIP Transit Softswitch. They have 25 Direct Destinations. Besides their Premium grade A-Z Service, Carrier grade service is also ideal for the Retail service providers all over the world including Middle East.

Currently Golden Telecom is running its operation from Dhaka office with four core departments,

- Network Operations Center (NOC)
- Billing
- Rates
- Routing

Figure 2.1 Regional Office Network of GT

Besides these some others departments like Accounts, Sales and Marketing are also working to keep this business stable for Golden Telecom. They consistently update the list to ensure low priced wholesale routes. Golden Telecom operates some of these routes directly. Before dealing with the customers all the routes are manually tested and strictly monitored to assure best quality possible.

Three types of products are available in Golden Telecom, Platinum, Premium and Economy. Quality varies with each plan. Each route has three or more redundant carriers. All these products includes 24/7 real time CDR traffic monitoring and standard customer care from expert support team.

VoIP carriers always look for fast interconnection and flexible payment methods. Wholesale voice is most popular because of this. Wholesale voice is commonly used by the providers offering PC to Phone, Calling card and callback services. Payment method is simple. Customers can pay through Credit Card, Paypal, Cash deposits and online banking.

2.2 Technical Components

For terminating voice traffic Golden Telecom mainly uses SIP protocol and codec G.711, G.723.1 and G.729. They use MVTS II routing engine to run switching, routing and billing operations. MVTS II is a wholesale VoIP platform ALOE Systems.

Some key functions of MVTS II manifesto are:

- MVTS II provides security for actual time value which entitles the carriers to persuade their business benefit. This feature assures standard ASR and ACD.

- A key feature of MVTS II is that it naturally combines profitable and non-profitable routes for producing favorable ASR.

- The route choosing system of MVTS II permits every vendor or customer specialization. It allows carriers to provide various products based on profit level, price, time, priority, quality and many other specifications or their blending.
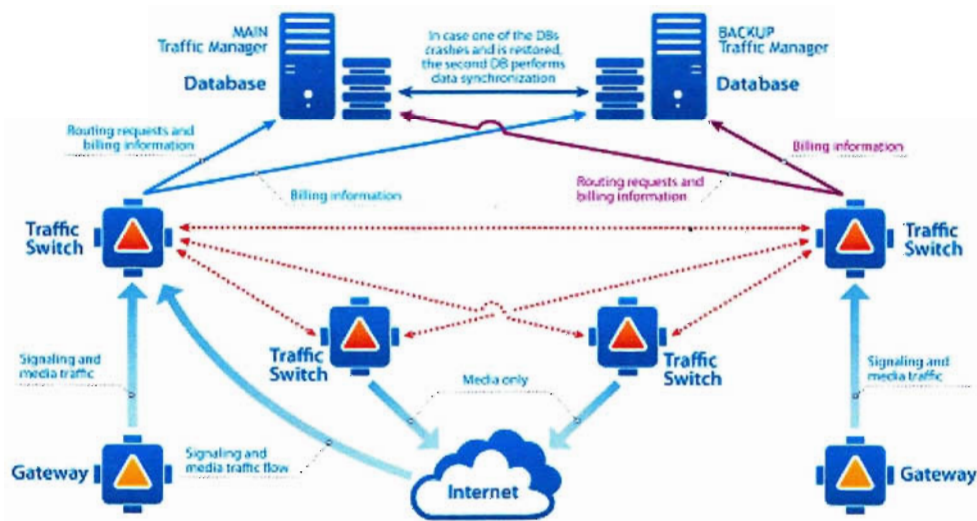

Figure 2.2 Architecture of MVTS II

2.2.1 Functional Layers of MVTS II

MVTS II has two functional layers,

1. The switching layer which is incorporated with one or more Traffic Switches (TS).
2. The management layer which is referred to as Traffic Manager (TM).

2.2.1.1 Switching Layer

Traffic Switches administer H.323 and SIP calls. They execute bipartisan transformation of voice CODECs and signaling protocols when needed. Every Traffic Switch is a comprehensive SBC which allocates a firm connectivity alloy with emerging partners and end users, secretes the network topology, guarantees safety from DoS attacks and allows interoperability of multi-vendor equipment. Moreover, each Traffic Switch is the main source of call statistics that is inspected and visualized by means of the Traffic Manager.

Individual Traffic Switch incorporates the following functional modules, each module being an impartial software application:

Signaling Node: SN issues bipartisan transformation of H.323 and SIP signaling protocols. It also provides load balancing between media junctions based on the contemporary CPU load of the server hosting every signaling node. Any number of media nodes can work with one signaling node.

H.323 Gatekeeper and SIP Registrar: They allow load sharing and manage registration appeals between the signaling nodes. The applicable data is redirected to the Traffic Manager when a customer attempts to register to the System. Based on the acknowledgement received from the Traffic Manager the registration appeal is either accepted or rejected. Load balancing is based on the ongoing call per second rate of every signaling node.

Management Node: MN gives unified control over other effective modules, stores entire configuration data, and performs as a collection juncture for the system stats.

Media Node: Media node execute transformation of voice CODECs, manage media flows and work as an RTP media proxy. In an MVTS II system the amount of media nodes required rely on the predicted number of concurrent call sessions that imply RTP media proxy operation.

Traffic Manager Node: TMN guarantees interaction between the Traffic Switch and Traffic Manager by means of DBMS users. Traffic Manager Node also supervises Interconnection with ENUM registers and external billing system, codec categorizing and distributing calls between termination devices within equipment groups.

Command Line Node: CLM is a telnet server by using any telnet client which authorizes clients to log on to a switching host.

Synchro Node: Synchro node confirms synchronization of the system operates. It also ensures control over the used facilities. The synchro node interacts with the signaling and traffic manager nodes.

Traffic Switch functions: Some functions of Traffic Switch are given below:

1. Traffic Switch handle H.323 v2 - v4 and SIP (RFC3261) calls and it bipartisan transformation of the signaling protocols.
2. It converts voice CODECs like G.729, G723.1, G.711A/U, GSM FR, Speex, iLBC, AMR.
3. RTP media proxy options.
4. Traffic Switch passes H.323 and SIP video using H.261, H.263 CODECs.
5. Traffic Switch ensures intensified dependability and almost limitless extensibility of the System.
6. Command-line interface for system administration.

Hardware and Software Requirements: Traffic Switch is developed to run on Debian GNU or Linux Lenny 5.0 (32-bit). Due to the distributed architecture of the switching layer the hardware prerequisites for the platform intended to host the system totally rely on the projected system capability and the selected method of redundancy.

Capacity: The following performance figures are given for deployments where the signaling and media nodes run on separate servers with Dual PIV 3.0 GHz/ 2 GB RAM with 1 Gbps networking interfaces.

Signaling node:

- Up to 3500 concurrent call sessions

- Traffic growth rate up to 50 new calls per second

- PDD: 30 ms and below

Media node (when installed on a dedicated server):

- Up to 1000 calls during proxy operation without codec conversion

- From 150 to 500 calls with codec conversion

- Media traffic latency from 10 to 100 ms depending on the operation mode

SIP registrar/balancer and H.323 gatekeeper/balancer:

- Up to 1000 simultaneous registrations

- Up to 100 new registrations per second

- Registration handling latency: 500 ms and below   [Ref - 3]

2.2.1.2 Traffic Management Layer

The principal component of MVTS II is Traffic Manager. Traffic Manager transfers authentication and authorization of VoIP equipment. It also performs call routing, call analysis, validation and transformation of call numbers, traffic load sharing. Moreover, Traffic Manager performs QoS control functions and pre-billing.

TM incorporates two components:

Database: Database is a ground of ORACLE™ technology. The Database is a storehouse of required data for call routing and analysis of statistics.

Web Server: WS produces a appropriate graphical alloy for administration functions and customer access.

Traffic Manager Capabilities: Traffic Manager provides an advanced call routing mechanism, reporting tools and detailed network analysis. The exclusive architecture of Traffic Manager ensures limitless extensibility and suppleness to overcome fault-tolerance and redundancy.

- Traffic Manager makes a record of optional routing paths contemplating the termination costs and service quality grounded on the destination number. The destination number traffic prefix determines the termination alternatives.

- Traffic Manager finds the lower cost routing paths.

- Operators can apply specialized routing policies by using Traffic Manager.

- Traffic Manager provides adaptable customer authentication and authorization supervision.

- By using the configured number transformation patterns Traffic Manager translates the source and destination numbers.

- Traffic Manager is responsible for traffic overload evaluation and load sharing.

- It Keeps various statistical parameters (number of calls, ASR, ACD, QoS, etc.) for dial peers and gateways.

- Vendors can change the customer rate plans if needed by using Traffic Manager.

- Traffic Manager represents various operational statistics such as peak and current System loads, total number of calls, ACD, total calls duration, ASR, QoS, etc.

- Calculates dynamic statistical parameters as EMA and allows configuration of the EMA period.

- Traffic Manager also Supports MNP, inter/intrastate routing and other particular attributes.

Traffic Manager Specifications:

- Average response latency: 60 ms.

- Supported dial peers (traffic termination opportunities): 10 mln and above.

- Traffic growth rate (new calls per second): 70.

- Number of authentication objects: not limited.

- Authentication rate (authentications per second): 300 and above with the total population of ID records amounting to 1,750,000.

- Authentication and authorization latency: 60 ms and below with the total number of ID records 2 mln.

- Number of users simultaneously accessing the web-server: 1000.

- Number of destinations in a customer-facing rate plan: not limited.

- Number of identified equipment per one vendor: not limited.

Hardware and Software Requirements:

The Traffic Manager Database software runs on the following operating systems:

- SUSE Linux Enterprise Server (SLES) 10
- Red Hat Enterprise AS/ES 5
- Red Hat Enterprise AS/ES 4
- Red Hat Enterprise AS/ES 3

Recommended browsers:

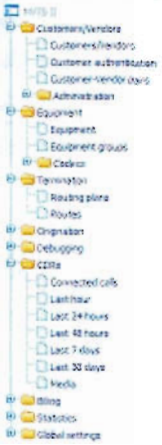- Microsoft Internet Explorer v. 8
- Mozilla Firefox v. 3
- Opera 9

The following server specification is capable of supporting the traffic growth rate of up to 70 calls per second with the number of configured dial peers of up to 10'000'000:

- Double AMD Opteron or Double Xeon CPU 3.0GHz
- 8Gb RAM
- RAID 0+1 of 4 15000rpm SCSI disks
- 1Gb Ethernet   [Ref - 3]

Figure 2.3 MVTS II Softswitch Interface

2.2.2 Quintum Gateway

Quintum Tenor AXT2400 Gateway is used in Golden Telecom for providing support to analog endpoints and PSTN. This gateway includes 24 FXO ports, echo cancellation and support for improved transmission.

Figure 2.4 Quintum Gateway

SIP based system and more suitable networks are supported by Quintum Gateway. To run small or medium-sized business Quintum Gateway is the absolute option. The Quintum Tenor AXT2400 gateway is simple to manage with a web-based GUI. To take VoIP to the next level it carries built in attributes such as gatekeeper, intelligent call routing, QoS and IVR support.

Specifications:

- **Telephony Interfaces:** 24 x FXO
- **Auxiliary Ports:** 10/100 Ethernet, RJ-45, RS-232
- **Concurrent Calls:** up to 24
- **Protocol Support:** DHCP, DNS, H.323, SIP, SNMP, TCP, UDP
- **SIP Endpoints:** up to 60
- **Chassis Type:** 1U form factor
- **Echo Cancellation:** yes
- **Manageable:** yes

In last few years Golden Telecom formed business relations with many carriers around the world such as BTCL (Bangladesh), Airtel (India), Tata Communications (India), Reliance (India), PTCL (Pakistan), Vodafone (Belgium), TeliaSonera (Sweden), Telenor etc.

2.3 Physical View

Figure 2.5 NOC Workstation

Figure 2.6 Billing Workstation

**Figure 2.7 Routing Workstation**

Figure 2.8 Conference Room

Chapter 3

VoIP Overview

3.1 VoIP

VoIP stands for Voice over Internet Protocol. VoIP technology enables telephone calls to be made across computer networks such as Internet. In VoIP analog voice signals are converted into digital data packets. To make calls using VoIP we need internet connection, VoIP carrier and ideal computer audio system. Some carriers use common telephones and unique adapters to connect to the network. VoIP calls are lot cheaper than our long distance GSM calls. The principal drawbacks of VoIP are short duration calls and cheapen voice quality. The subscribers commonly face this problem when the traffic load is huge.
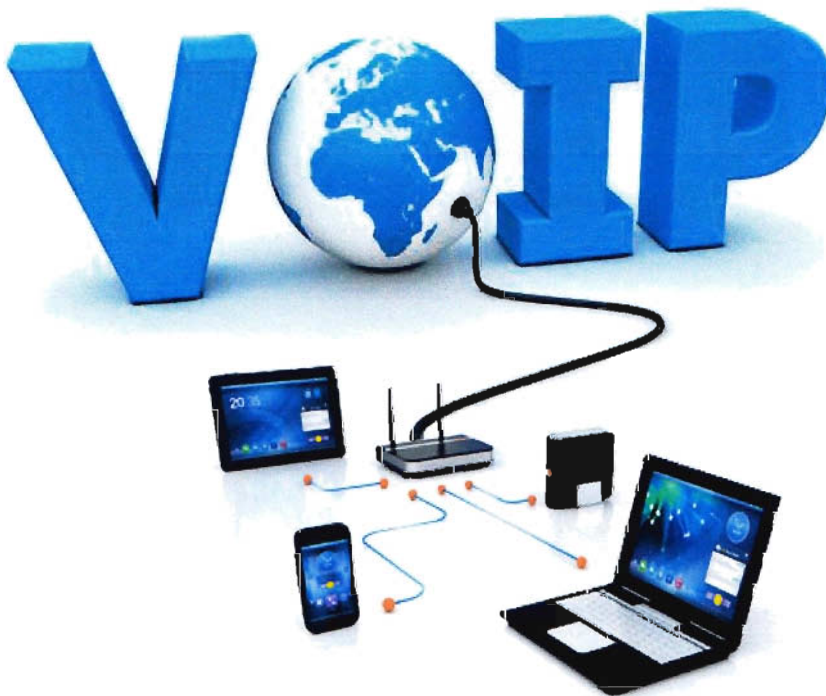
Figure 3.1 VoIP Technology

We have found some reasons that are key to the development of VoIP business:

1. Low-priced calls.
2. Add-on services and merged messaging
3. Combination of data and voice infrastructures.

There are 3 different ways to place a VoIP call:

ATA: The easiest and most familiar way is along with the use of a device known as ATA, Analog Telephone Adaptor. For the use with VoIP, Analog Telephone Adaptor allows to connect a ideal telephone to computer or Internet connection. The ATA converts analog signal to digital. After taking the analog signal the ATA converts it to digital data and then transmits the data over the internet.

Figure 3.2 ATA Device

Its using procedure is too simple. The telephone cable needs to be connected to the ATA and thus the device is ready to place VoIP calls. In VoIP service ATA devices are often locked when sold so that it cannot be used by any compatible provider.

IP Phones: The IP Phones looks like normal analog telephones. This unique phone is built with cradle, buttons and a handset just like the traditional telephones. But IP Phones have RJ-45 Ethernet connector where the normal phone has the RJ-11 phone connectors. IP Phones are directly connected to the routers. These devices have built in hardware and software for handling the VoIP calls. An IP Phone delivers superior quality audio performance.

Figure 3.3 Different IP Phones

IP Phones uses various control protocols like SIP, SCCP etc. An IP Phone supports many features like Email, contacts, easy sharing of contacts.

Some hardware elements of an IP Phone:

→ Key pad to enter phone number and text.

→ Display to show caller ID and messages.

→ Speaker and microphone.

→ General Purpose Processor.

→ DACs & ADCs.

→ A Digital signal processor to process RTP messages.

→ Ethernet or wireless network hardware.

Computer to Computer: This is surely the simplest way to place calls in VoIP. For long distance calls in this process there is almost no payment required. Some company offers free or very cheap software that can be used to make free VoIP calls. To use this process some other components are required such as speakers, a microphone, good internet connection and a sound card. Whatever the distance is, Computer to Computer calls are totally free of cost. Users only have to pay the bandwidth fees to the ISP companies.

Figure 3.4 Computer to Computer

3.2 VoIP Protocols

There are many signaling protocols that are used in VoIP services. When audio or video data are transmitted between two ends, different protocols must be applied to look for the appropriate path through which data will be transmitted between ends. These protocols are known as call signaling protocols.

The most popular signaling protocols are H.323 and Session Initiation Protocol (SIP). The functions of H.323 and SIP are too similar but they differ in design where H.323 is a binary protocol and SIP is an ASCII based protocol. But the real fact is that both H.323 and SIP can do the same thing which is totally invisible to their users. So practically the differences between H.323 and SIP are peripheral.

3.2.1 H.323

In 1995 when researchers were looking to find a way to establish communication between two computers to send and receive media files, H.323 came into focus for the first time. This is International Telecommunication Union's Standard. Vendors should follow this standard when giving VoIP service to their customers. At the beginning it was developed for multimedia conferencing purpose. After few years it was extended for VoIP service. It was first launched in 1996. The 2nd edition of H.323 came into focus in 1998.

3.2.1.1 Components of H.323

Terminals: Terminals are said to be the LAN user destination points. They assure real time, two way communications. H.245, Q.931, RAS and RTP is required to be supported by all H.323 terminals. H.245 enables the usage of the channels, Q.931 performs call signaling and call setup, RTP carries voice packets and RAS is used for interacting with the gatekeeper. H.323 terminals may also include T.120 data conferencing protocols, video CODECs and support for MCU. A H.323 terminal can communicate with either another H.323 terminal, a H.323 gateway or a MCU.

Gateways: H.323 gateway acts as an endpoint on the network. It generates real-time, bipartisan transformation between H.323 terminals on an Internet Protocol network and other ITU Networks. More specifically, H.323 works as a translator that translates various transmission structures, i.e. from H.225 to H.221. An H.323 can also perform translation between audio and video CODECs. The gateway is the interface between the PSTN and the Internet. They take voice from circuit switched PSTN and locate it on the public Internet and vice versa. Terminals in a single LAN can communicate with each other directly where gateways are optional in this regard. The terminals communicate via gateways using the H.245 and Q.931 protocols when the terminals on a network need to communicate with an endpoint in some other network.

Gatekeepers: The most important element of H.323 standard is the gatekeepers. They perform the obligations of a manager. Gatekeepers provide support to the registered clients. They perform like a middle point for all calls within their zone. Some of the functionalities that gatekeepers provide are listed below:

Admissions Control: Gatekeepers are capable of either accepting or rejecting admission depending on call authorization, source and destination addresses.

Call Management: The gatekeeper may maintain a list of ongoing H.323 calls. This information may be necessary to indicate that a called terminal is busy, and to provide information for the Bandwidth Management function.

Address Translation: They translate a false address to transport address. To perform this action updated translation table is used.

Call Signaling: Gatekeepers are capable of accomplishing call signaling with the destination points and can handle the call signaling itself. They can also govern the endpoints to connect the Call Signaling Channel directly to each other.

Call Authorization: In the case of not getting proper authorization gatekeepers can deny calls from a terminal by using H.225. This can result a restriction over terminals for some time periods.

Bandwidth Management: Control of the number of H.323 terminals permitted simultaneously access to the network. Through the use of H.225 signaling, the Gatekeeper may reject calls from a terminal due to bandwidth limitations.

Multipoint Control Units: The fundamental endpoint MCU generates the vision to 3 or more terminals and gateways to engage in a multipoint conference on a network. The MCU consists of a mandatory MC and optional MP. The MC determines the common capabilities of the terminals by using H.245 but it does not perform the multiplexing of audio, video and data. The multiplexing of media streams is handled by the MP under the control of the MC.
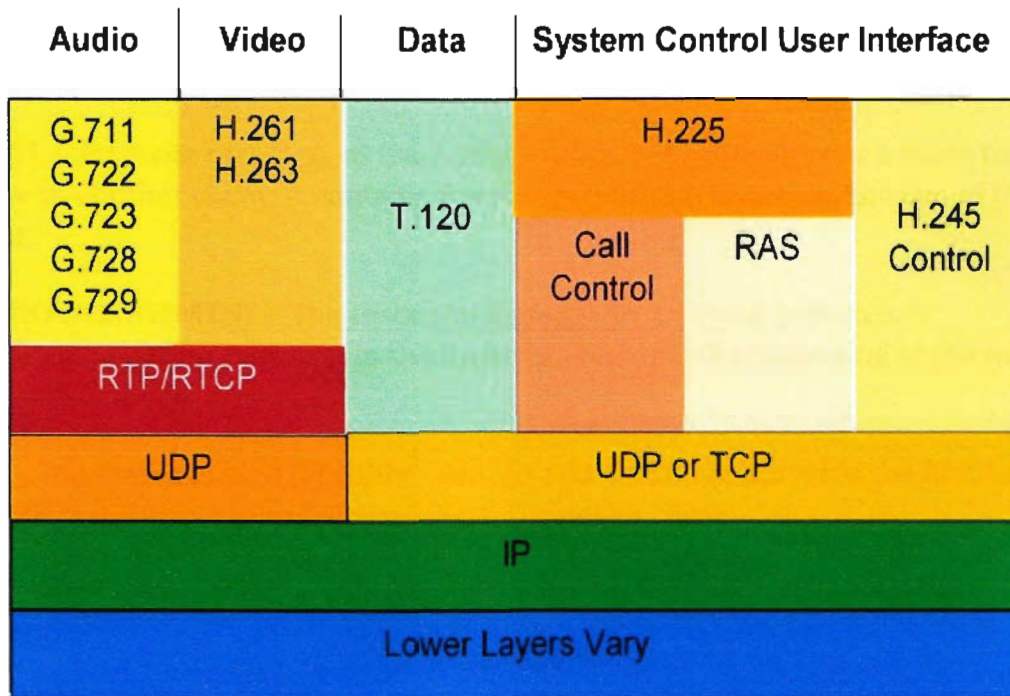
3.2.1.2 Call Setup in H.323

The call set up process in H.323 is given below:

- Finding a gatekeeper that takes the management of that endpoint.
- Registration of the endpoint with its gatekeeper.
- Endpoint enters the call setup phase.
- The capability exchange takes place between the endpoint and the gatekeeper.
- The call is established.
- When the endpoint is ready, it can terminate the call. Gatekeeper can also initiate termination of calls.

As a system identification H.323 defines some ITU-T and IETF protocols. These are:

→ Real-time Transport Protocol (RTP)
→ H.225.0 Registration, Admission and Status (RAS)
→ H.225.0
→ H.235
→ H.239
→ H.245
→ H.450
→ H.460

Audio	Video	Data	System Control User Interface		
G.711 G.722 G.723 G.728 G.729	H.261 H.263	T.120	H.225		H.245 Control
			Call Control	RAS	
RTP/RTCP					
UDP		UDP or TCP			
IP					
Lower Layers Vary					

Table 3.1 H.323 Protocol Stack

3.2.2 SIP

SIP was also developed in 1995. SIP is IETF's standard for initiating VoIP connections. SIP is an application layer control protocol. The structure of SIP protocol is too similar to the design of Client-Server Protocol. In SIP client generates a request and sends it to the server. After processing the request the server sends an acknowledgement back to the client.

SIP protocol has ACK and INVITE messages. These messages describes the procedure of a reliable path though which call control messages will be processed. SIP protocol never depends on TCP for authenticity. SIP relies on the Session Description Protocol for conveying the messages for codec identification.

3.2.2.1 SIP Components

The SIP standard has two components: User Agents and Network Server.

3.2.2.2 SIP Messages

SIP describes many messages which are used for the communication between the client and the server. They are:

INVITE – An invite message, as the name implies, is a request from a client to speak to another client. It contains the media type and other capabilities of the client.

ACKNOWLEDGEMENT – The response message to an invite message. It represents the final message in the invite process and the beginning of the media exchange (voice).

BYE – The message sent by either client to end a call. The server is the first to receive the BYE message followed by the opposite client.

OPTIONS – Allows the client to collect information on other clients and the servers.

CANCEL – Cancels any message exchanges that are in progress but not yet completed.

REGISTRATION – Registers a client with a server and allows the client to use the services on the network.

3.2.2.3 SIP Operation

SIP Addressing: A SIP URL like sip:username@host recognizes the SIP hosts. A single host or a whole group can be classified by a SIP address.

Locating a SIP server: Commensurate to the URL, a client can either send the request directly to the IP address and port or can send the request to a SIP proxy server.

SIP Transaction: Once the host part of the Request URI has been resolved to a SIP server, the client can send requests to that server. A request together with the responses triggered by that request makes up a SIP transaction.

SIP Invitation: A successful SIP invitation consists of two requests: a INVITE followed by ACK. The INVITE request asks the caller to join a particular conference or establish a two party conversation. After the caller has agreed to participate in the call, the caller confirms that it has received that response by sending an ACK request. The INVITE request contains a session description that provides the called party with enough information to join the session. If the caller wishes to accept the call, it responds to the invitation by returning a similar session description.

Locating a User: A callee may keep changing its position with time. These locations can be dynamically registered with the SIP server. When the SIP server is queried about the location of a callee, it returns a list of possible locations. A Location Server in the SIP system actually generates the list and passes it to the SIP server.

Changing an Existing Session: Sometimes we may need to change the parameters of an existing session. This is done by re-issuing the INVITE message using the same Call ID but a new body to convey the new information.
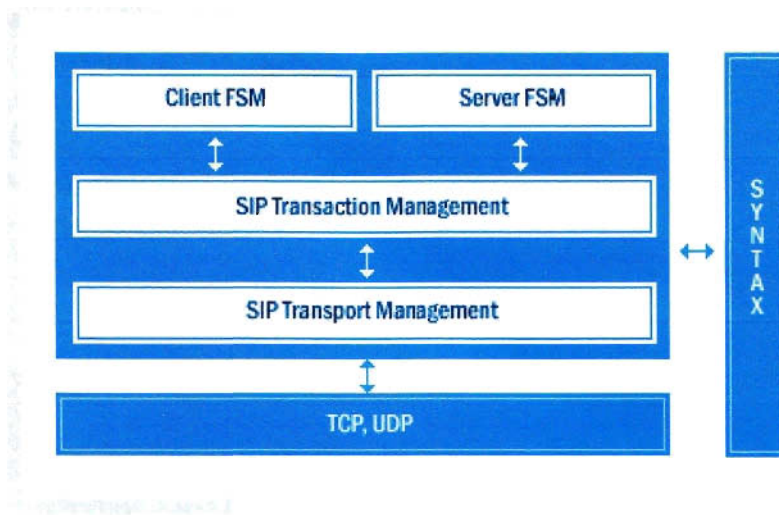
Table 3.2 SIP Protocol Stack

3.2.3 Comparison between H.323 and SIP:

H.323	SIP
It's a complex protocol	SIP is comparatively simpler
It requires full backward compatibility	It doesn't require full backward compatibility
H.323 is Binary based	ASCII based
H.323 not very scalable	SIP is highly scalable
It has hundreds of elements	It has only 37 headers
Loop detection is difficult in H.323	Loop detection is comparatively easy in SIP
It has complex signaling	It has simple signaling
H.323 is not very modular	SIP is very modular

Table 3.3 Comparison between H.323 and SIP

3.2.4 Supporting Protocols

There are some supporting protocols also. They are:

→ Media Gateway Access Protocol
→ Real Time Streaming Protocol
→ Session Description Protocol
→ RTP and RTCP
→ Session Announcement Protocol
→ Resource Reservation Protocol

3.2.5 CODECs

The most common voice CODECs includes G.711, G.723, G.726, G.728, and G.729. A brief description of each CODEC follows.

G.711 –Converts voice into a 64 kbps voice stream. This is the same CODEC used in traditional TDM T1 voice. It is considered the highest quality.

G.723.1 – Two different types of G.723.1 compression exist. One type uses a CELP compression algorithm and has a bit rate 5.3 kbps. The other type uses a Multi Pulse – Maximum Likelihood Quantizer algorithm and provides better quality sound. This type has a bit rate of 6.3 kbps.

G.726 – Allows for several different bit rates, including 40, 32, 24, and 16 kbps. It works well with packet to PBX interconnections. It is most commonly deployed at 32 kbps.

G.728 – Provides good voice quality and is specifically designed for low latency applications. It compresses voice into a 16 kbps stream.

G.729 – One of the better voice quality CODECs. It converts voice into an 8 kbps stream. There are two versions of this CODEC, G.729 and G.729a. G.729a has a more simplified algorithm over G.729

3.2.6 Voice Quality Metrics

CODEC: The choice of CODEC is the first factor in determining the quality of a call. Generally, the higher the bit rate used for the CODEC, the better the voice quality. Higher bit rate CODECs, however take up more space on the network and also allow for fewer total calls on the network.

Network: The biggest factor in call quality is the design, implementation, and use of the network that the voice calls are riding on. A typical VoIP call will start on a LAN at a CPE, go through a WAN connection to a provider cloud, and then go back out the other end. The CPE LAN and WAN links are most vulnerable to overuse and errors. Most VoIP quality issues are typically caused at these links.

Packet jitter: Jitter caused by changes in the inter-arrival gap between packets at the endpoint. The packets should arrive evenly spaced to allow for a seamless conversion into analog voice. If the packet gap changes, the user could experience degradation in quality. If the packet gap gets sufficiently large, the phone's packet jitter buffer will not be able to wait for the late packet, and the phone will drop the late packet. There are three different types of packet jitter such as RFC jitter, instantaneous jitter and absolute jitter.

Packet loss: The actual loss of voice packets through a network. Packet loss is often caused by congestion at one or more points along the path of the voice call or by a poor quality link (one that experiences physical layer errors).

Delay: Sometimes referred to as envelope delay, refers to the time it takes for the voice to travel from the handset of one phone to the ear piece of the other phone. Envelope delay is the sum of the delay caused by the CODEC of choice, jitter buffer in the phone, and the path time it takes for the packets to get through the network. A large delay can make conversation difficult.

Echo: Echo is a common problem for VoIP networks. It is important to note that, unlike packet jitter, packet loss, and delay, echo is not caused by the IP network. Echo is an analog impairment. It is extremely difficult to passively monitor for echo. The best way to detect echo is by placing a call onto the network with a known "good" device or capturing the voice packets of a call and playing them back for analysis.

There are two types of echo on analog voice networks – hybrid echo and acoustic echo. Hybrid echo is generated by impedance mismatches at various analog or digital points on the network. The most common location for the generation of hybrid echo is at a 2-wire to a 4-wire conversion point. Acoustic echo is generated at the phone. It occurs when the voice leaving the speaker is picked up by the microphone.

Chapter 4

VoIP in Bangladesh

4.1 Inauguration

Government of Bangladesh wanted to establish a legal route for international voice call by using VoIP technology for the mass people at affordable price. BTRC published the Regulatory and Licensing Guidelines for VoIP Service on July 22, 2012. It was developed with envision of ILDTS Policy, 2010.

The Licensing process of VoIP service provider is an open Licensing process. The regulatory commission issues new Licenses to the applicants if required depending on the term and conditions. The duration of a VoIP License is five years. The Licensee needs to renew the License after the first term. The License renewal duration is 2 years and it includes fees specified by the Government.

Fees:

Application Fee (non refundable)	Tk. 5,000.00 only
License Acquisition Fee	Tk. 5,00,000.00 only
Annual License Fee	Tk. 1,00,000.00 only
Audited Gross Revenue Sharing with the Commission	20% of call rates
Performance Bank Guarantee	Tk. 2,50,000.00 only

Charges:

International call rates are decided by the commission. After deducting VAT the prevailing international incoming call rates in Bangladesh Taka (BDT) shall be shared in the following proportion:

1. VSP keeps 15% of the call rates.
2. VSP pays to IGW(s) 15% of the call rates.
3. VSP pays to ICX(s) 15% of the call rates.
4. VSP pays to ANS 20% of the call rates.
5. VSP pays to the Commission 35% of the call rates.

4.2 Guidelines

To provide an overview of the licensing and regulatory infrastructure and to encourage local businesses in Telecommunication sector BTRC published some guidelines in 2012. We will mention some important parts of these guidelines.

4.2.1 General Requirements

The following are the principal legal statutes governing the telecommunication industry in Bangladesh:

- The Bangladesh Telecommunication Regulation Act, 2001 (as amended).

- The Wireless Telegraphy Act, 1933 and The Telegraph Act, 1885, for matters which are not covered by the Bangladesh Telecommunication Regulation Act, 2001 (as amended).

- The Bangladesh Telecommunication Regulatory Commission (Licensing Procedure) Regulations, 2004 (as amended).

- The Bangladesh Telecommunication Regulatory Commission (Interconnection) Regulations, 2004.

- Any Act of Parliament or ordinance and the Regulation(s) made or to be made by the Commission.

Applicant(s) shall be disqualified from obtaining a license for any of the provisions listed in points below applies to its owner(s) or to any of its directors or partners or to the applicant(s) himself if:

- He is an insane person and/or he has been sentenced by a court under any law, other than this Act, to imprisonment for a term of 2 (two) years or more, and a period of 5 (five) years has not been elapsed since his release from such imprisonment.

- He has been sentenced by a court for committing any offence under the Act and a period of 5 (five) years has not been elapsed since his release from such imprisonment.

- He has been declared bankrupt by the Court and has not been discharged from the liability of bankruptcy.

- He has been identified or declared by the Bangladesh Bank or by the court or by a bank or financial institution as a defaulter loanee of that bank or institution.

- Any of his Licenses has been cancelled by the Commission at any time during the last 5 (five) years.

- Any proceeding is going on against the applicant(s) or its owner(s) or any of its directors or partners for violation of the Act / Any Regulations / Rules / Guidelines / Bye-laws / Directives / Instructions / Orders / Circulars / Decisions of the Commission etc.   [Ref - 5]

4.2.2 System and Services

- The Licensee will provide services for international incoming calls.

- Existing IGWs and ICXs are considered to be the FBO and VSP operators are considered to be NFBO for handling voice traffic.

- Voice calls shall be routed (terminating to Bangladesh) to VSPs through licensed IGWs.

- All VSP shall establish and maintain interconnection among its own IGWs and ICXs. VSPs will develop and maintain interconnection facilities to connect ICXs.

- The Licensee shall have to submit the status of bandwidth utilization from time to time as directed by the Commission.

- The Licensee shall have sufficient monitoring facilities of their bandwidth utilization.

- The Licensee shall enter into end to end Service Level Agreements (SLA) with overseas carriers and will negotiate regarding fees for call termination by themselves within the provision of directives from the Commission, if there is any. Agreed fees and SLA(s) with the overseas Carrier, IGW and ICX must be submitted to commission and shall be vetted by the Commission.

- The Licensee shall have the system arrangement to identify, block and eliminate illegal traffic with the assistance of the respective IGWs from which VSP shall be receiving the bandwidth for call termination.

- The licensee shall provide necessary information, where applicable to the Commission and to the National Monitoring Centre (NMC) for online and offline monitoring and analyzing of Call Detail Record (CDR), Traffic Data, Log files, Call Accounting, Signaling, Quality of Service (QoS) and other systems as may be directed by the Commission from time to time with the assistance of the respective IGWs from which VSP shall be receiving the bandwidth for call termination.

- VSP must provide Calling Line Identification (CLI) for any voice call.

- If the licensee has different types of licenses, the licensee shall have separate physical infrastructure and location for each type of services. The licensee shall not be allowed to use same switching system for different types of services.

- Corporate private IP Networks (e.g. VPN) are within the scope of the regulatory framework in that they are subject to the authorization directives by the Commission.   [Ref - 5]

4.2.3 Eligibility

- VSP operator license will be issued only to Bangladeshi entities (Resident Citizens, Non-Resident Bangladeshi, proprietorships, partnerships-under the Partnership Act, 1932 and companies registered under 'Joint Stock Companies and Firms' under the Companies Act, 1994). Foreign entities (citizens, companies or subsidiaries or holding companies) shall not be eligible to be Owners / Directors / Shareholders / Investors / Partners of these licensee entities.

- One business entity will be allowed to get only one VSP License.

- A person / company / firm shall not be eligible to apply for VSP license, if there are any outstanding dues owed by him / it or any of its directors / partners to the Commission.   [Ref - 5]

4.2.4 Cancellation, Suspension of License and Fines

The Commission may cancel, suspend the License and impose fine and the Licensee shall be liable for action as per Bangladesh Telecommunication Regulation Act, 2001(as amended) on the following ground including but not limited to –

- That any information furnished in the Application form for obtaining the License is found incorrect/ false.

- That required fees and charges are not paid by the Licensee as per the terms and conditions of the VSP License.

- That any share of it is transferred or issued without prior written permission of the Commission.

- That any of the conditions of the VSP License is violated.

- That the Licensee has ceased to carry on business mentioned in this license.

- That the licensee has disclosed or is involved with the disclosure of any information to anybody/ performing any illegal activities that may hamper National Security, Integrity, Sovereignty, Stability, and Harmony.

- That the Licensee is liquidated, bankrupt or insolvent that an application for declaration of bankruptcy or similar declaration or order is filed by the Licensee itself or a third party against the Licensee.

- That the Licensee violates or purports to violate any sections / terms and or conditions under the Act / Any Regulations / Rules / Guidelines / Bye-laws / Directives / Instructions / Orders / Circulars / Decisions of the Commission etc.   [Ref - 5]

4.2.5 Tariffs and Pricing

The Licensee shall, before selling or leasing capacity or providing facilities, like providing access, data center, co-location, etc. submit to the Commission in writing:

- A tariff chart / schedule containing the maximum and minimum charges that it proposes to charge for such purposes, and its justification for the charges.

- The description of the sale or lease and facilities, the terms and conditions and all other relevant information which it proposes to publish to its customers. The information to be published must be in a form that is readily available, current and easy to understand.

- Application forms for all types of capacity sale and for providing facilities.

The Licensee shall obtain the written approval of the Commission before making any changes to the approved tariffs and charges.   [Ref - 5]

4.3 ICX

ICX stands for Interconnection Exchange. ICX refers to switching systems which provides interconnections among the existing/future telecommunication network of the operators and allows monitoring, Lawful Interception (LI) facilities and Roaming Number Portability. Here is a list of some leading Interconnection Exchange Operators in Bangladesh:

1. BTCL
2. Bangla ICX Ltd.
3. Agni ICX
4. CloudTel
5. GAZI Networks Ltd
6. Sheba
7. Bangla Telecom Ltd

4.4 IGW

IGW stands for International gateways. IGW refers to switching systems through which international voice traffic is sent and received. IGW allows physical monitoring of the traffic flow and allows monitoring, Lawful Interception (LI) facilities and Roaming Number Portability. Some leading IGWs of Bangladesh:

1. Mir Telecom
2. Bangla Trac Communications Limited
3. BTCL
4. Novotel Limited.
5. DBL Telecom Limited

4.5 IIG

IIG stands for International Internet Gateways. IIG refers to switching systems through international data traffic is sent and received. There are more than 20 IIGs in Bangladesh. BTCL, Mango Teleservices Limited, Aamra Companies, Fiber @ Home Limited are the leading IIGs of Bangladesh Telecommunication field.

Chapter 5

Network Topology of Golden Telecom
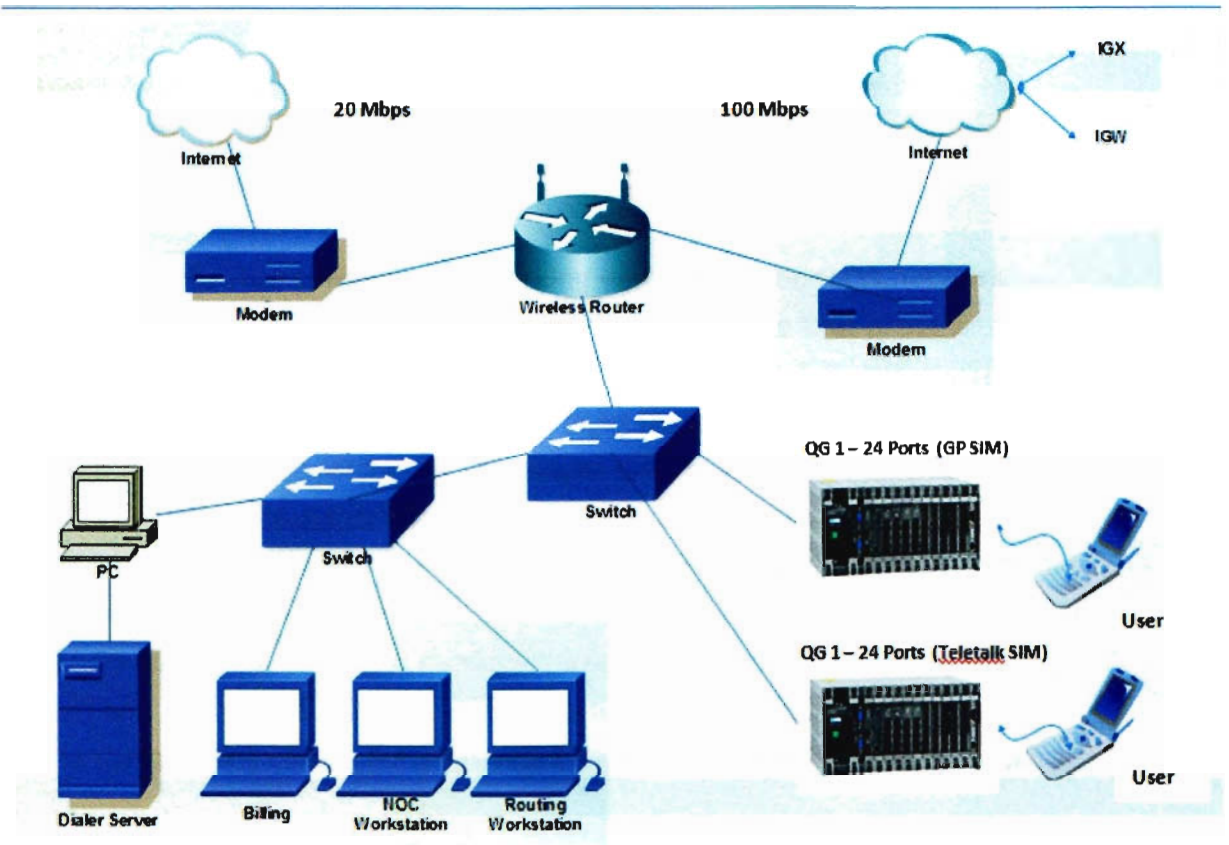
Figure 5.1 Network Topology of Golden Telecom

Chapter 6

VoIP in Golden telecom

6.1 Networking Devices

Router: A router is a networking device whose software and hardware are usually tailored to the tasks of routing and forwarding information. For example, on the Internet, information is directed to various paths by routers.

Switch: A Switch is a computer networking device that connects network segments. The term commonly refers to a Network bridge that processes and routes data at the Data link layer (layer 2) of the OSI model. Switches that additionally process data at the Network layer (layer 3 and above) are often referred to as Layer 3 switches or Multilayer switches. The term switch does not generally encompass unintelligent or passive network devices such as hubs and repeaters. The first Ethernet switch was introduced by Kalpana in 1990.

Wireless Modem: A wireless modem is a modem that bypasses the telephone system and connects directly to a wireless network, through which it can directly access the Internet connectivity provided by an Internet service provider (ISP).

Quintum Gateway: See **2.2.2**

Table 6.1 IP Address/Blocks

POP/Link	IP Address/Block
Golden Telecom POP	10.1.2.0/24
Amra POP	10.1.1.0/24
Mango POP	10.1.3.0/24
Amra & Golden Telecom Link	10.1.12.0/24
Mango & Golden Telecom Link	10.1.23.0/24
Dialer Server	10.1.2.43
Billing Workstation	10.1.2.24
NOC Workstation	10.1.2.25
Routing Workstation	10.1.2.26
Quintum Gateway 1	10.1.2.47
Quintum Gateway 2	10.1.2.48

6.2 BGP Configuration

Amra
R1
10.1.1.1/24

R3
10.1.3.3/24
Mango

10.1.12.0/24

10.1.23.0/24

R2

10.1.2.2/24
Golden Telecom

as 200

as 100

as 300
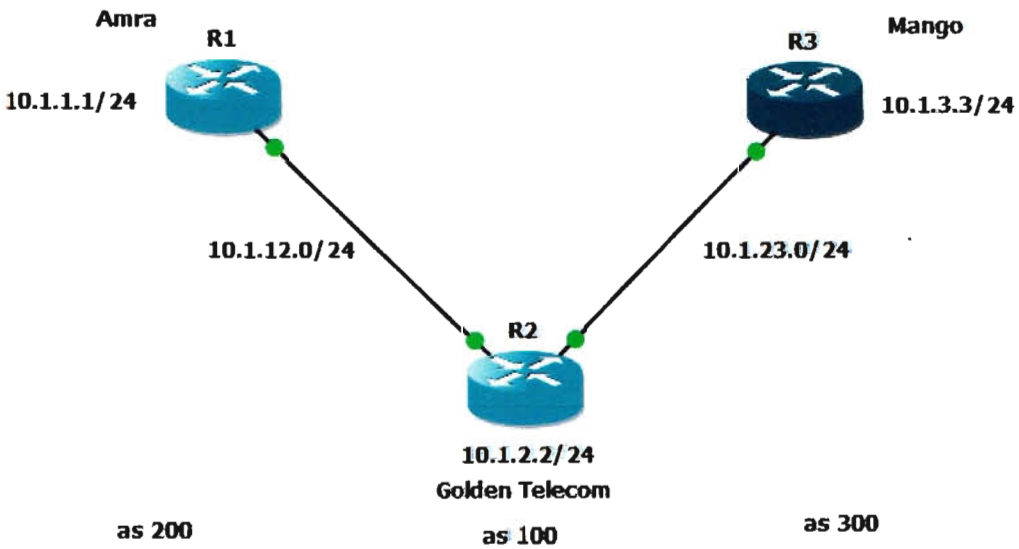
Figure 6.1 BGP Configuration

Router 1 (Amra) Configuration:

```
hostname Amra
boot-start-marker
boot-end-marker
no aaa new-model
no ip icmp rate-limit unreachable
ip cef
no ip domain lookup
ip tcp synwait-time 5
interface Loopback0
ip address 10.1.1.1 255.255.255.0
interface FastEthernet0/0
ip address 10.1.12.1 255.255.255.0
duplex auto
speed auto
interface FastEthernet0/1
no ip address
shutdown
duplex auto
speed auto
router bgp 200
no synchronization
bgp router-id 10.1.1.1
bgp log-neighbor-changes
network 10.1.1.0 mask 255.255.255.0
neighbor 10.1.12.2 remote-as 100
no auto-summary
no ip http server
no ip http secure-server
control-plane
gatekeeper
```

```
shutdown
line con 0
exec-timeout 0 0
privilege level 15
logging synchronous
stopbits 1
line aux 0
exec-timeout 0 0
privilege level 15
logging synchronous
stopbits 1
line vty 0 4
login
end
```

Router 2 (Golden Telecom) Configuration:

```
hostname Golden
boot-start-marker
boot-end-marker
no aaa new-model
no ip icmp rate-limit unreachable
ip cef
no ip domain lookup
ip tcp synwait-time 5
interface Loopback0
ip address 10.1.2.2 255.255.255.0
interface FastEthernet0/0
ip address 10.1.12.2 255.255.255.0
```

```
duplex auto
speed auto
interface FastEthernet0/1
ip address 10.1.23.2 255.255.255.0
duplex auto
speed auto
router bgp 100
no synchronization
bgp router-id 10.1.2.2
bgp log-neighbor-changes
network 10.1.2.0 mask 255.255.255.0
neighbor 10.1.12.1 remote-as 200
neighbor 10.1.23.1 remote-as 300
neighbor 10.1.23.3 remote-as 300
no auto-summary
no ip http server
no ip http secure-server
control-plane
gatekeeper
shutdown
line con 0
exec-timeout 0 0
privilege level 15
logging synchronous
stopbits 1
line aux 0
exec-timeout 0 0
privilege level 15
logging synchronous
stopbits 1
line vty 0 4
login
end
```

Router 3 (Mango) Configuration:

hostname Mango
boot-start-marker
boot-end-marker
no aaa new-model
no ip icmp rate-limit unreachable
ip cef
no ip domain lookup
ip tcp synwait-time 5
interface Loopback0
ip address 10.1.3.3 255.255.255.0
interface FastEthernet0/0
ip address 10.1.23.3 255.255.255.0
duplex auto
speed auto
interface FastEthernet0/1
no ip address
shutdown
duplex auto
speed auto
router bgp 300
no synchronization
bgp router-id 10.1.3.3
bgp log-neighbor-changes
network 10.1.3.0 mask 255.255.255.0
neighbor 10.1.23.2 remote-as 100
no auto-summary
no ip http server
no ip http secure-server
control-plane
gatekeeper

shutdown
line con 0
exec-timeout 0 0
privilege level 15
logging synchronous
stopbits 1
line aux 0
exec-timeout 0 0
privilege level 15
logging synchronous
stopbits 1
line vty 0 4
login
end

Load Sharing:

Golden Telecom bgp 1
no synchronization
bgp log-neighbor-changes
bgp bestpath as-path multipath-relax
network 10.1.2.0 mask 255.255.255.255
neighbor 10.1.12.1 remote-as 200
neighbor 10.1.12.1 route-map BGP2 in
neighbor 10.1.23.1 remote-as 300
neighbor 10.1.23.1 route-map BGP in
maximum-path 5
no auto summary

route-map BGP permit 10

set metric 0

set local-preference 90

set origin incomplete

route-map BGP 2 permit 10

set metric 0

set local-preference 90

set origin incomplete

set as-path prepend 3

6.3 Configuration of Quintum Tenor AXT2400 Gateway

At first we have launched the configuration manager by clicking the **START** button from **Programs** menu.
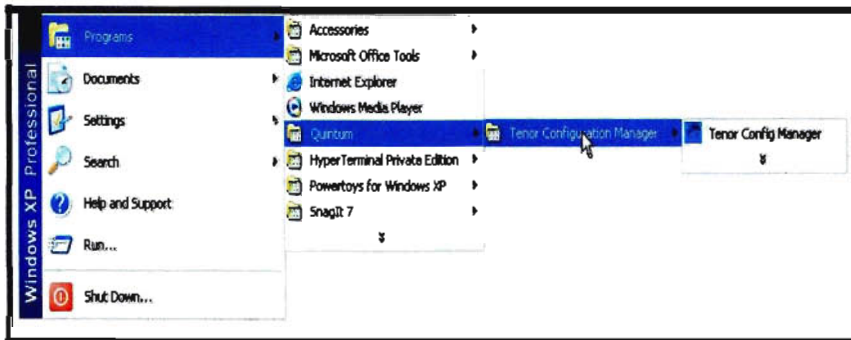
Figure 6.2 Configuration Manager Software
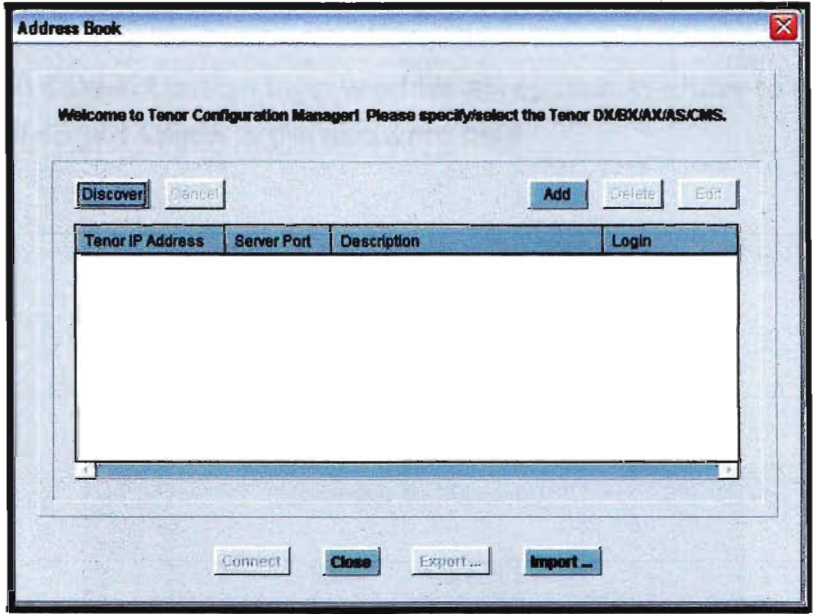
Figure 6.3 Tenor Configuration Manager Window

After that we have to click on the **Discover** button. The Configuration Manager will search the local network for active gateways and display the results.
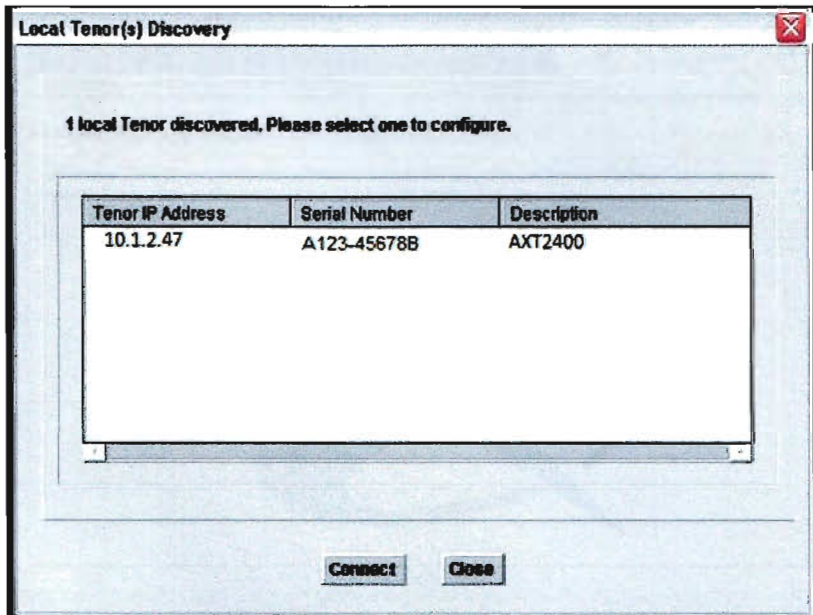
Figure 6.4 Discovery Window

By clicking on **Connect** button login window will appear. We have to enter **admin** in the login field and **admin** in the password field.

Figure 6.5 Login Window

Then we have to click **OK** button. It will open the Quintum Configuration Wizard.

Figure 6.6 Quintum Configuration Wizard

We click the **Next** button to proceed. In the next window we have to specify whether to use DHCP or a static IP.
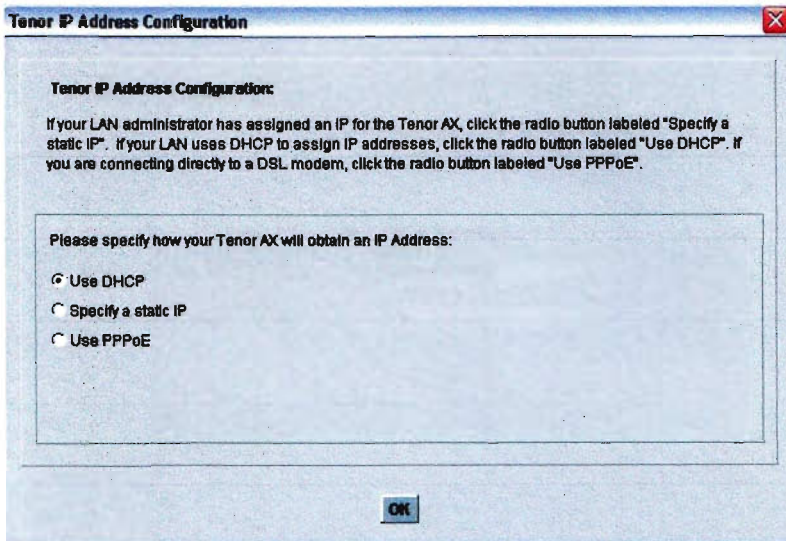
Figure 6.7 IP Address Configuration Window

In this step we have to specify whether the gateway should use DHCP obtained or manually configured DNS servers, and then we click the **OK** button.
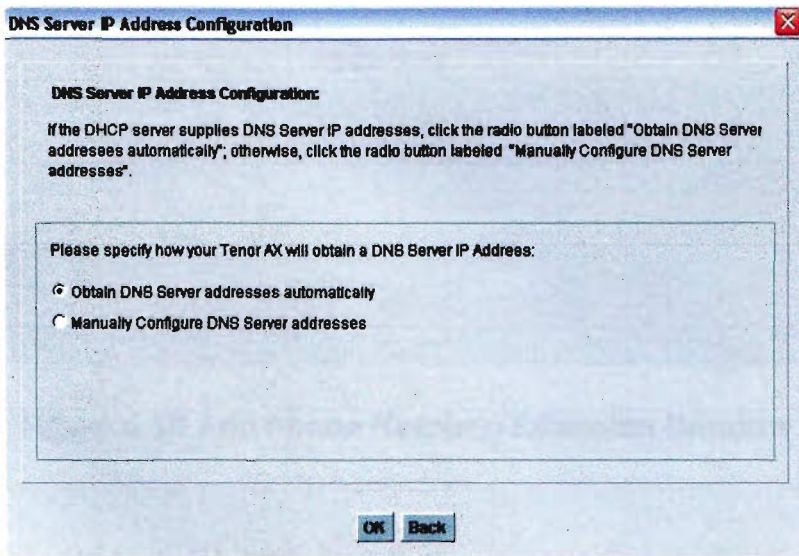
Figure 6.8 DNS Server IP Address Configuration

In the next step a dial plan configuration window opens and we will click **Skip.**
We will click next in the following step and phone port configuration screen
appears.
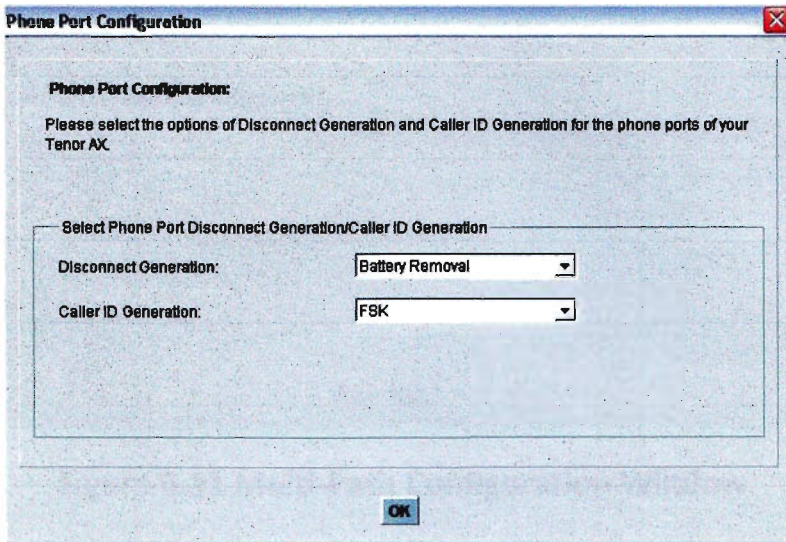
Figure 6.9 Phone Port Configuration Window

Here we will click **OK** and will add a number in the next phase.
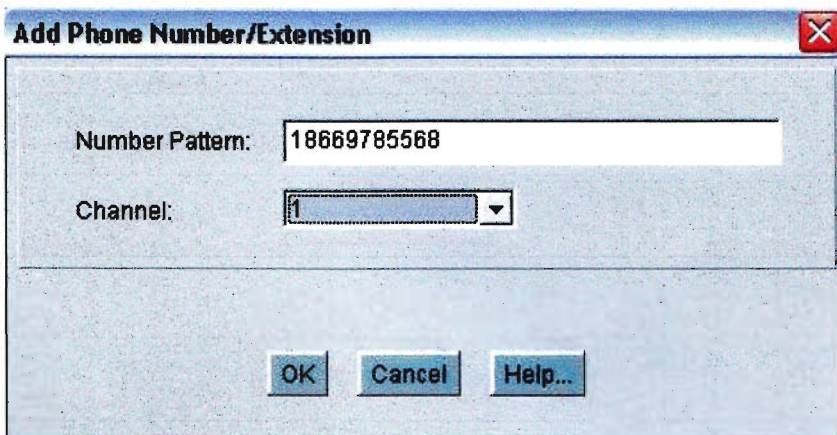
Figure 6.10 Add Phone Number/Extension Window

We will choose NO and click **OK** button in the next window to disable the Pass
Through feature.
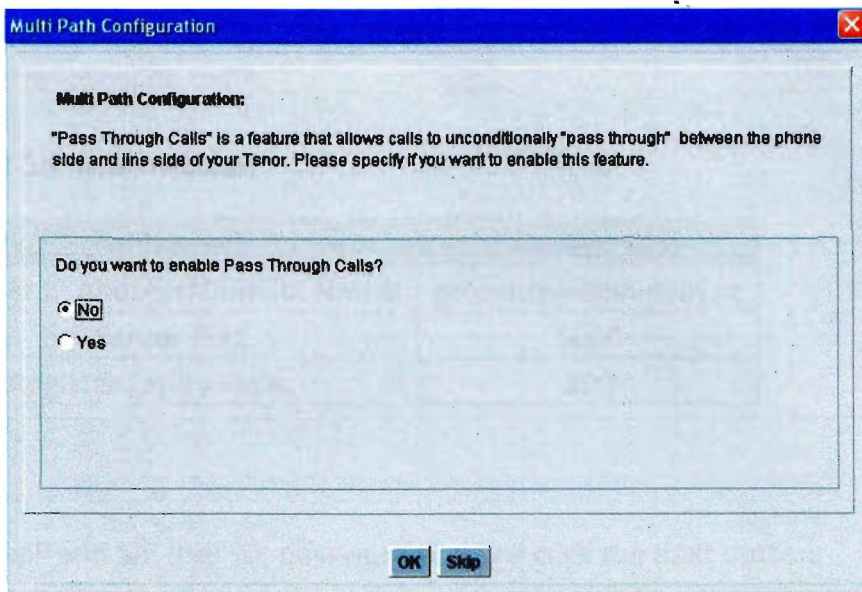
Figure 6.11 Multi-Path Configuration Window

In the VoIP gateway configuration window we chose SIP only and click **OK** and continue the configuration process.
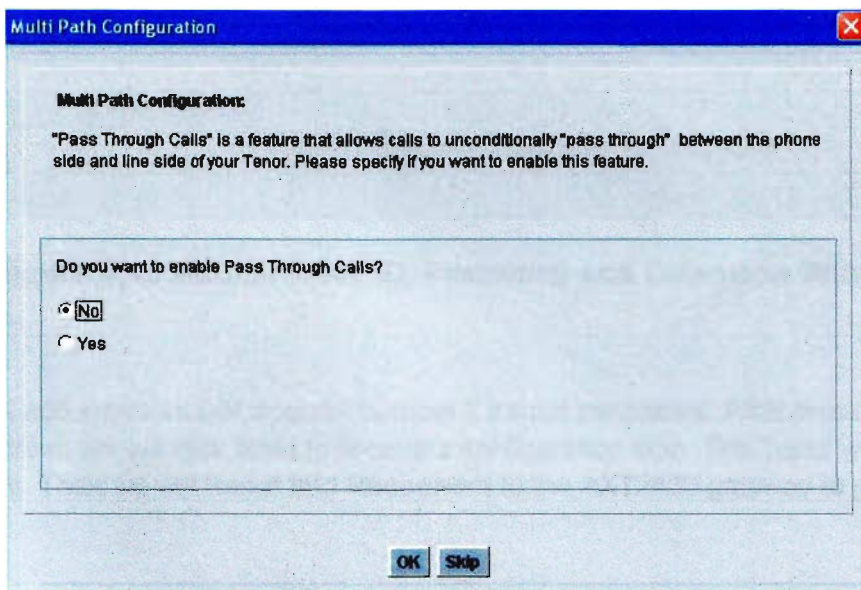
Figure 6.12 VoIP Gateway Configuration Window
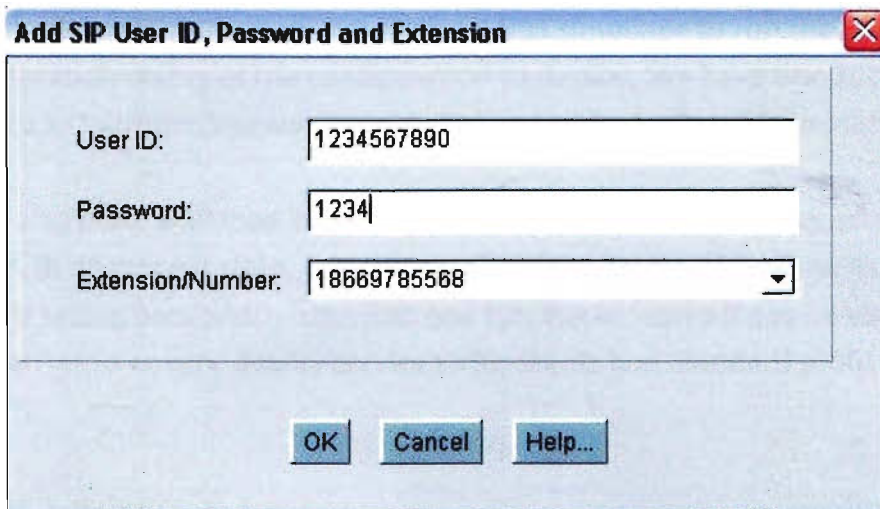
When the SIP server configuration screen appears we will enter the information as it is in the following table.

Table 6.2 SIP Information

Field Name	Entered Value
SIP Server IP Address/Domain Name	goldentelecom.com.sg
SIP Server Port	5060
Register Expiry Time	300

Then we will add SIP user ID, password and will click the next button.

Figure 6.13 Add SIP User ID, Password and Extension Window

We will not add anymore SIP account number if it's not necessary. After proceeding to the next screen we will click **Next** to finish the configuration step. The Tenor will save all the settings. Then we will **Reset** and **Reconnect** to the AXT2400 gateway to get started.

Summary

Network management has various aspects. This thesis has focused on configuration management. The objective of this thesis was to configure a highly extensible network for Golden Telecom including a Quintum Gateway also. After finishing the execution and testing the performance it can be said that the objective of this thesis has been achieved. However there are many areas for development in this regard.

In this thesis we have studied different aspects of Voice over Internet Protocol, network configuration management and combined in a network management system. Structures of Golden Telecom and different protocols of VoIP were studied in this project. Then we have used GNS3 simulator to run the unit tests to assure operation ability of the configuration to deploy. We have also successfully configured a Quintum Gateway which will be used for VoIP call termination.

We have also done BGP load balancing for Golden Telecom by using network simulator. In its current state, this part is not suited to productive use as Golden Telecom is taking bandwidth from just one ISP. But in future it can be very helpful for this carrier to ensure quality service to its clients and standard profit level.

List of Figures

List of Tables

List of Abbreviations

SBC - Session Border Controller

EMA - Exponential Moving Average

MNP - Mobile Number Portability

RAS - Registration Admission Status

RTP - Real Time Transport Protocol

VoIP - Voice over Internet Protocol

VPN - Virtual Private Network

WAN - Wide Area Network

CDR - Call Detail Record

SIP - Session Initiation Protocol

MVTS - Multiprotocol VoIP Transit Softswitch

ASR - Answer-Seizure Ratio

ACD - Average Call Duration

TS - Traffic Switch

TM - Traffic Manager

DoS - Disk Operating System

CPU - Central Processing Unit

RTP - Real-time Transport Protocol

DBMS - Database Management System

ENUM - Telephone Number Mapping

GSM - Global System for Mobile

AMR - Adaptive Multi-Rate

RAM - Random Access Memory

PDD - Post Dial Delay

QoS - Quality of Service

MNP - Mobile Number Portability

SLES - SUSE Linux Enterprise Server

RAID - Redundant Array of Inexpensive Disks

SCSI - Small Computer System Interface

PSTN - Public Switched Telephone Network

FXO - Foreign Exchange Office

GUI - Graphical User Interface

IVR - Interactive Voice Response

BTCL - Bangladesh Telecommunications Company Limited

PTCL - Pakistan Telecommunication Company Limited

NOC - Network Operations Center

ATA - Analog Telephone Adaptor

SCCP - Signalling Connection Control Part

DAC - Digital-to-Analog Converter

ADC - Analog-to-digital converter

ISP - Internet Service Provider

ASCII - American Standard Code for Information Interchange

MCU - Multipoint Control Units

ITU - International Telecommunication Union

LAN - Local Area Network

MP - Multipoint Processors

MC - Multipoint Controller

IETF - Internet Engineering Task Force

UDP - User Datagram Protocol

TCP - Transmission Control Protocol

ACK - Acknowledgement

URI - Uniform Request Identifier

RTCP - RTP Control Protocol

TDM - Time Division Multiplexing

CELP - Code Excited Linear Prediction

PBX - Private Branch Exchange

CRTP - Compressed RTP

PPP - Point-to-Point Protocol

HDLC - High-Level Data Link Control

CPE - Customer Premises Equipment

ILDTS - International Long Distance Telecommunication Services

BTRC - Bangladesh Telecommunication Regulatory Commission

VSP - VoIP Service Provider

ANS - Access Network Service

FBO - Facilities Based Operator

SLA - Service Level Agreements

NMC - National Monitoring Centre

CLI - Calling Line Identification

ICX - Interconnection Exchange

IGW - International gateways

IIG - International Internet Gateways

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