5.0 VOIP SOLUTION:

The proposed Unified Communication (UC)/ VOIP system should provide Three Tier architecture.

Tier-I: VoIP Exchange with Unified Communication (UC) features shall be installed at scope Minar Delhi. The System at Delhi will act as Master Unit with VoIP and Unified communication features. Bidders are required to include all the Hardware (various Servers) and softwares to meet the requirements.

Tier-II: The VoIP Systems at Asset Head quarters will have VoIP servers along with Gateways for integrating with Legacy Exchanges. These systems primarily equipped with SIP based VoIP features, however will utilize the UC facilities installed at Central site Delhi (Tier-I).

Tier-III: Remote units shall have VoIP end points which includes phones and Gateways. These systems primarily equipped with SIP based VoIP features, and under the control of Tier-II and utilize the UC facilities installed at Central site Delhi (Tier-I).

5.0.1 COMMUNICATION TECHNOLOGY

The solution shall form a Single System Image to station users and system administrators across all locations with distributed architecture installed at Assets. There should be an option to implement as Single Image or Distributed Architecture without foregoing the features of UC and VoIP. The distributed solution must be able to support distributed locations using WAN links for call control signals and voice communications signals. The system should be SIP based VoIP and standards-based principles of IP Multimedia Subsystem to allow core UC/ VoIP services to be delivered to each user in the enterprise.

The VoIP system and associated Subscriber Station equipment, Network Management & Simple CDR information systems shall be provided in a single package. The required quantities at various Assets/Basins are specified elsewhere in the document.

The IP Unified Communication System shall have ability to run TDM and IP on the same platform using same software based on Servers and Gateway architecture. The system should support IPv4 & IPv6. The system should be capable of supporting Analogue, Digital, IP Telephones, PC Based IP Softphone with suitable client software.

5.0.2 OPEN STANDARDS

For easy integration/interoperability with third party applications the offered IP Unified Communication System Should use open standards for its Operating system, Call processing, signaling & networking. There should not be any Proprietary/ vendor specific protocols, Call processing, signaling & networking.

5.0.3 OPERATING SYSTEM

Preferably should be based on open standard platform. The O/ S should be hardened to avoid any virus attacks, and provide Security in various aspects.

5.0.4 RELIABILITY /SURVIVABILITY

I. System should be fully redundant with two Servers. The two servers should work in Active / standby or Active-Active mode i.e. if one Server fails the second server
should be able to take the complete load of the calls automatically (without any
manual intervention) without dropping any existing calls.

II. There should be no restriction on number of endpoints being backed up in case of
one sever failure. The System should be capable of supporting following:

a) Duplicating the Power Supply units for the Servers.

b) The offered system should be capable of Hot Swapping of servers without switching
off the system, where the necessary Server can be interchanged or replaced even
in online conditions without disruption of services.

c) The offered IP PBX and UC software should be a modular and fault tolerant
architecture with self-healing capabilities

d) The system software should be capable of incorporating error detection/ correction,
system recovery and alarm escalation paths

e) The system software should have a watchdog monitor to continuously observe and
report on health and sanity of the applications, the base OS and the critical
environmental conditions

f) During Server change over, administrator should be able to do basic administration.

g) In case of WAN link failure, the remote locations should be able to do basic calling,
i.e. extension to extension call/ extension to trunk call. This functionality should be
provisioned for the gateways at remote sites also.

h) Geographical survivability feature must be supported for the communication system
at the core (Tier-I) to provide reliable, fault-tolerant, critical availability. It should be
able to provide service from geo-redundant locations at up to 1 second of round trip
delay across the network.

5.0.5 SECURITY

a) For enhanced security IP Unified Communication System should be able to encrypt
the IP calls end to end with appropriate securities. There should not be any
limitation in terms of IP endpoint type.

b) Access to the system should be secure .Access mechanism like SSH, HTTPS, SNMP v3 should be used.

c) It shall be possible to bar unauthorized user to connect to the system.

d) System should also provide the ability for encrypted audio streams from IP Phone
to the multi-party conference bridge.

e) Authorization Codes: Every user shall have his own authorization code to make
outgoing calls thereby ensuring no misuse of the system. System shall give the user
complete flexibility to dial his Personal identification number or authorization codes.

f) The offered solution should support forced entry of account codes for every call
made by the users. However there should be provision to force entry of account
codes for calls made on a specific trunk or to a specific telephone number.

g) Authentication, Authorization, and Accounting Services- System should have
support for (AAA) services whereby it should allow to store and maintain
administrator account information on a central server. System should support account information being stored on an external AAA server or locally on the system itself. Both types of accounts may be used at the same time.

h) System should be able to provide security violation alert on some predefined endpoint i.e. analog/digital/IP Phone by a pre-recorded announcement. The system should monitor and report the following types of security violations:

- Login violations
- Remote access barrier code violations
- Authorization code violations
- Station security code violations.

5.0.6 SYSTEM ADMINISTRATION

The proposed communications solution should support advanced administration functions using user friendly software tools. The management system should be operated using GUI tools, formatted screens, pull down menus, valid entry choices, templates, transactions, scheduling, and database import/ export. The required hardware & software tools used to access and perform administrative operations shall be included. The administration shall be local for Exchanges at Assets/ Basins, and also shall have Centralized management at

I. CENTRALIZED MANAGEMENT

a) It is required that all management/maintenance operations must be supported by a single management system with a unified customer database for all station users across all locations.

b) All station moves/ adds/ change operations must be able to be implemented transparently across all locations. System monitoring, diagnostic, and maintenance operations for all locations must be supported using a single centrally located applications server at the HQ location with distributed client workstations across the network.

c) The core elements and management software should present a single enterprise wide view of the solution’s routing policies.

II. ASSET/ BASIN WISE MANAGEMENT

a) CONFIGURATION / MANAGEMENT

- Configuration Manager (GUI) for configuration of remote gateways, end points.
- Alarm monitoring, call monitoring and CDR monitoring
- SNMP v2 Agent
- Command Line Interface

b) CALL MANAGEMENT FEATURES

- Automatic call type detection: Voice/Modem/Fax
- Answer and Disconnect Supervision
- User programmable dial plan support
- Automated load balancing
• Forced IP routing and IP port mapping
• Automatic appending and stripping of digits to dialed numbers
• Call Detail Records
• Call Blocking
• Caller ID delivery

c) SYSTEM MAINTENANCE TERMINAL

System administration & maintenance Terminal with the above capabilities and tools, including standard maintenance terminal screen features and functions shall be provided with each Server. All the maintenance process should be able to be carried out using the web interface.

5.0.7 SYSTEM DIAGNOSTICS

I. The proposed communications system is able to support advanced diagnostics and maintenance capabilities. The diagnostics must be able to monitor and detect system failures and errors. At minimum the following diagnostic functions must included:

a) Monitoring of processor status
b) Monitoring and testing of all port and service circuit packs
c) Monitoring and control of all power units
d) Emergency transfer and control of active processing, switching, and power systems to backup systems
e) Originate alarm information and activates alarms.
f) System should have in-built health monitoring software to check the functioning of the system.
g) The system should have in built diagnostic features such as isolation/ detection of fault in the system/ server/ junction and restoration of faulty system/junctions after rectification.
h) When the system go down due to any reason, it should have auto restart capability to automatically reload the system software after system power is restored/fault rectified.

II. REMOTE DIAGNOSTICS

a) The system shall provide Remote diagnostics of Servers, Gateways and end points. It shall provide full access to system diagnostic functions to monitor system performance and perform necessary maintenance operations. The interface shall be web based.

5.0.8 ALARM REPORTING

a) The proposed communications system solution must alert maintenance personnel of system errors and failures by a variety of means, including an audible alarm, management screen alert, and printed record.
b) System should able to define threshold values, and measure generated alarms. The solution must be able to be programmed to define customized major and minor alarm conditions. Alarms shall be recorded in an Alarm Record Log. The devices shall include: tone receivers; DTMF senders and receivers; conference bridges; trunk routes etc.

5.0.9 ENDPOINTS

a) The Unified Communication System must be able to support all type of endpoints i.e. analog phones, digital phones, IP Phones with appropriate hardware. IP Unified Communication System must have the support for IP Soft phones as well.

b) The IP endpoints spread across all locations should have a desktop controlled IP endpoint.

c) The desktop client should be able to initiate a point to point SD video call to any other location.

d) The desktop client should have an inbuilt IM and presence functionality

5.0.10 INTERFACE COMPATIBILITY

The telephony server should support following trunk interfaces/ protocols with appropriate hardware/ software:

a) CO trunks-system should be able to provide interface to the CO trunks with appropriate hardware.

b) ISDN-PRI with QSIG: The server should support ISDN-PRI trunks.

c) H323 and SIP trunks: The server should have the integrated H323 and SIP trunk Interface based on Internet Protocol (IP) technology to provide a lower cost of usage by transmitting voice over corporate Intranet, or private Local Area Network (LAN).

5.0.11 FLEXIBLE DIALING PLAN

It is required that the proposed solution shall support flexible multi-digit dialing up to 10 digits plan between stations across all locations. The system should support one enterprise dial plan across all multi-vendor systems.

5.0.12 CALL PROCESSING CAPACITY

The call processing capacity of the proposed system must be able to satisfy current and future communication requirements. The minimum call processing capacity of the proposed primary controller is 500,000 Busy Hour Call Completions (BHCCs).

5.0.13 SCALABILITY:

The proxy should be available in Microsoft Windows and Linux versions. The proxy should be able to scale to 40000 users.
5.0.14 CALL FEATURES

Call features required to operate transparently across the system locations must at minimum include:

- Call Transfer
- Call Forwarding – All Calls
- Call Forwarding – Don’t Answer
- Call Forwarding – Busy
- Automatic Callback
- Calling Number & Name on Telephone Display
- Trunk ID on Telephone Display
- Call Diversion Information on Telephone Display
- Add-on Conference
- Call Waiting
- Barge-in (Busy Override)
- Emergency Access to Attendant
- Paging System Access
- Station User Roaming (Logical Station Assignment)
- Message Waiting Activation

System features required to operate transparently across the system locations must at minimum include:

- Automatic Camp-on
- Automatic Recall
- Automatic Alternate Routing
- Meet-Me-Conferencing
- Trunk Callback Queuing
- Hunting
- Uniform Call Distribution
- Call Detail Recording
- Uniform Dial Plan

5.0.15. NETWORKING PROTOCOL - QSIG

System shall provide Industry standard based QSIG functionality on ISDN-PRI trunks

Call Services

- Basic Call Setup
- Name and Number Transport
- Transit Counter
- Called/Calling/Busy/Connected Name and Number
- Name Identification Services
- Diversion (Call Forwarding)
- Diversion (Call Forwarding) with Reroute (using Path Replacement)
- Call Transfer
- Call Offer
- Path Replacement
- Call Transfer into QSIG Message Center

5.0.16. IN-BUILT CONFERENCING

The Unified Communication System should be capable of supporting multiple 6-Party audio conferencing in any combination of external and internal calls. The conferences should be password protected.
5.0.17. USER-BASED ACCESS CONTROL:
End-users may log onto the proxy and change their personal preferences. They should be able to set the redirection to home, out of office and mailbox. They should also be able to see the call list from the proxy, initiate conferences from the proxy and start call-back services.

5.0.18. SPEED DIAL:
The speed dial feature should allow easy translation of short telephone numbers into SIP URLs. The speed dial should be able to be set up on domain, group and user level.

5.0.19. FIND ME FOLLOW ME:
Ability to call several user agents in parallel or serial based on set probabilities

5.0.20. INSTANT MESSAGING:
Instant messaging must supported by the standard procedures of the proxy. In addition to this, the proxy should supports storage and forwarding of messages to users that are known but not registered with the proxy. A welcome message may be delivered upon the first registration of a user.

5.0.21. CALL HUNTING:
Unlimited call stages support for each stage, a specific waiting time and different ringing melody can be specified. Should allow round robin and one-way contact attempts. A default definition may specify a call-waiting queue for later processing.

5.0.22. ADDRESS BOOK:
Address book on per domain basis or per group or per user must be supported and possibility to directly import address books from Microsoft outlook must be supported.

5.0.23. MASS DEPLOYMENT
Mass deployment must be possible by ability to deploy VOIP end point devices from central site without having to physically configure each remote CPE.

5.0.24. ATTENDANT CONSOLE
An attendant console with general capabilities and features of attendant console applications shall be provided for each Server.
5.0.25. TELEPHONY SPECIFICATIONS

- Voice algorithms: G.723.1a, G.729ab, G.726 ADPCM, G.711 with auto negotiation
- Fax support: Industry standard T.38 and Group III at 2.4, 4.8, 7.2, 9.6, 14.4 kbps
- Modem over IP
- FXS/FXO support.
- Standard 50 pin Telco "D" Connectors
- Coding: A-law, µ-law
- Enhanced (Carrier Grade) Echo Cancellation: ITU Rec. G168, up to 128 msec tail size
- Loop Start, Reverse Battery, Battery Disconnect
- Tandem/TDM switching

5.0.26. IP NETWORK SPECIFICATIONS

- LAN Interface: 1 Fast Ethernet port (10/100/1000 Base-T) with standard RJ-45
- DHCP Client
- QoS Support: IP TOS, DiffServ

5.0.27. VOIP NETWORK SPECIFICATIONS

- H.323 v.4 Gateway, and Gatekeeper
- SIP User Agent
- Adaptive Voice Activity Detection (VAD) with Comfort Noise Generation (CNG)
- Adaptive Jitter Buffer
- Packet Loss Compensation
- Should work securely behind NAT enabled firewall
- Security: IP Filtering
- Packet Multiplexing technology for saving Bandwidth

5.0.28. SIP PROXY SPECIFICATION

Registrar and Location Server: Must be based on SIP (RFC 3261). Server should accept REGISTER requests. The register server should support authentication. Registrar server should offer location services.

Domain Hosting Support: Multiple Domains must be supported. Each domain should have its own log, its own user list, its own dial plan, its own registration policy, its own password, its own welcome message policy, and so on. DNS A, NAPTR, SRV, ENUM: should be supported.

User-Based Access Control: End-users may log onto the proxy and change their personal preferences. They should be able to set the redirection to home, out of office and mailbox. They should also be able to see the call list from the proxy, initiate conferences from the proxy and start call-back services.

Controlling: Proxy should generate controlling information based on patterns for an unlimited number of groups. The information should be possible to be accumulated over a period of three months and be able to be fed into standard tools such as Microsoft.

Instant Messaging: Instant messaging must supported by the standard procedures of the proxy. In addition to this, the proxy should supports storage and forwarding of messages to users that are known but not registered with the proxy. A welcome message may be delivered upon the first registration of a user.

Bidders shall submit brief write up on how these services are provided by their solution as part of the bid document and not mere compliance statements. These shall also be tested during SAT tests.
5.0.29. **VOIP EXCHANGE (MEDIA SERVER WITH MEDIA GATEWAY):**

The VOIP exchange (media server with media gateway) shall conform to the following minimum requirement.

<table>
<thead>
<tr>
<th>SN.</th>
<th>Parameter</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Media Server configuration</td>
<td>(1+1) hot standby mode</td>
</tr>
<tr>
<td>2</td>
<td>Server Capacity</td>
<td>New Delhi: Equipped with 500 User licenses, Expandable to 3000 IP users. Hardware at Delhi shall include to meet the UC functionalities as specified above. Nazira, Jorhat, Rajahmundry and Agartala &amp; Bokaro: Equipped with 500 User licenses, Expandable to 2000 IP users.</td>
</tr>
<tr>
<td>3</td>
<td>Hard disk(HDD)</td>
<td>200 GB or above</td>
</tr>
<tr>
<td>4</td>
<td>RAM</td>
<td>4GB or better</td>
</tr>
<tr>
<td>5</td>
<td>Mounting arrangement</td>
<td>19&quot; rack</td>
</tr>
<tr>
<td>6</td>
<td>Standard</td>
<td>H.323 and SIP dual stack, QSIG</td>
</tr>
<tr>
<td>7</td>
<td>Support Network Interface</td>
<td>PRI, E&amp;M, CO, FXO/FXS. Support local as well remote monitoring through LAN/WAN</td>
</tr>
<tr>
<td>8</td>
<td>Power</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Power Voltage</td>
<td>220V AC ± 10%,50Hz</td>
</tr>
<tr>
<td>9</td>
<td><strong>ENVIRONMENTAL</strong></td>
<td></td>
</tr>
<tr>
<td></td>
<td>a</td>
<td>Temperature</td>
</tr>
<tr>
<td></td>
<td>b</td>
<td>Humidity</td>
</tr>
</tbody>
</table>

5.0.30. **VOIP GATEWAYS, (ANALOG MODELS- FOR E&M/FXO/FXS EXTENSION):**

The VOIP gateway (analog models for E&M/ FXO/ FXS) shall conform to the following minimum specifications:

<table>
<thead>
<tr>
<th>S. No.</th>
<th>Parameter</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Mounting arrangement</td>
<td>19&quot; Rack mountable</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>---</td>
<td>---</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Number of ports</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>4 or 8</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Interfaces</td>
<td></td>
</tr>
<tr>
<td>a</td>
<td>E1</td>
<td></td>
</tr>
<tr>
<td>b</td>
<td>FXO/FXS voice interfaces-2 wire loop/ground start</td>
<td></td>
</tr>
<tr>
<td>c</td>
<td>E&amp;M: 4 wire and E&amp;M, Type-I to V</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Dialing</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>DTMF &amp; /Pulse</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Connectors</td>
<td></td>
</tr>
<tr>
<td>5</td>
<td>RJ-45 (E&amp;M), RJ-11(FXO/FXS)</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Voice Compression</td>
<td></td>
</tr>
<tr>
<td>6</td>
<td>As given above.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Protocols Support</td>
<td></td>
</tr>
<tr>
<td>a</td>
<td>H.323 and SIP dual stack, QSIG</td>
<td></td>
</tr>
<tr>
<td>b</td>
<td>Dynamic Host Configuration Protocol Networking (DHCP)</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Others</td>
<td></td>
</tr>
<tr>
<td>8</td>
<td>Echo cancellation, Jitter buffer</td>
<td></td>
</tr>
<tr>
<td></td>
<td>LAN</td>
<td></td>
</tr>
<tr>
<td>9</td>
<td>10/100 baseT or better</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Management</td>
<td></td>
</tr>
<tr>
<td>10</td>
<td>SNMP</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Power</td>
<td></td>
</tr>
<tr>
<td>11</td>
<td>220VAC, Single phase, 50Hz</td>
<td></td>
</tr>
</tbody>
</table>

### 5.0.31 HARD VOIP INSTRUMENT:

The VOIP instruments shall conform to the following minimum specifications:

a) IP telephone instrument must be fully compatible, hardware and software wise with the exchange

b) Hands-free dialing.

c) Push button dialing.

d) The hard phone should have full duplex Speakerphone with Echo cancellation.

e) IP telephone instrument should have two nos. of RJ-45 port, one for telephone and second for PC (RJ45) connectivity.

f) It should support DHCP based as well as statically configured IP Address assignment

g) It should be equipped with minimum 2X24 character LCD display

h) The phone should be IEEE 802.3af compliant for power needs and provisioning of suitable power adaptor

i) Standard based codec support for: G.711, G.722, G.729ab, G.726
5.0.32. **IP DESK VIDEO PHONE TYPE-I**

The Video Phone shall conform to the following minimum specifications:

a) Video desk phone should be based on open standards.

b) Video desk phone should support web browsing & email integration like outlook/lotus.

c) The video phone should work on its internal battery, in case of power failure providing 2hrs. backup.

d) 10” WXGA (1366x768) LCD display.

e) 5Mp HD video embedded camera; Auto focus, electronic shutter

f) Dual integrated microphones.

g) Two USB 2.0 host ports

h) Support for basic SIP features like conference, transfer etc as per SIP RFC 3261

i) 6 Party Call Conferencing (Ad hoc/click to conference)

j) Call Forwarding All Calls (activation/de-activation)

k) Call Forwarding Busy/Don’t Answer (activation/de-activation)

l) Extension to Cellular (mobile twinning)

m) Send All Calls (activation/de-activation)

n) Should be able to establish IM session with other users using XMPP ,p2p.

o) Should be able to do point to point video calling using standard p2p (720p HD,H.264)

p) Should be able to see the presence of other video phone users.

q) For better interoperability, the SIP video desk phone should be of the same make as the SIP based IP PBX.

r) Each SIP video endpoint in the system should be able to have at least 3 simultaneous registration points (in line with SIP’s subscription over registration principle), i.e. in case primary
gatekeeper fails, it should seamlessly get registered to the secondary gatekeeper and so on.

5.1. DESIGN CRITERIA FOR INTEGRATION WITH EXISTING SYSTEMS:

a) The system shall be interoperable with the existing LAN/ WAN of the organisation.

b) The Point-To-Multi Point and Point-To-Point radio based System at Assets/ Basins shall be integrated with the organisation’s Network Operations Centre (NOC) at Delhi, through SNMP to capture events generated by the Point-To-Multi Point and Point-To-Point radio based system over existing enterprise network.

c) Contractor should ensure compatibility of the systems supplied at different installations and Asset Centre with the exchanges and Networking infrastructure available at each of these locations.

d) Contractor should provide along with the technical bid the Integration schemes between the proposed communication equipment and existing exchanges, LAN/ WAN networks at base and remote locations.

e) Contractor should ensure compatibility of the supplied systems and equipment to different relevant Standards mentioned in the technical specifications. The technical proposal should highlight the relevant portion of the documents and catalogues of system, equipment and software conforming the same.

f) VOIP exchanges shall be considered for each Asset/ Basins for extending voice connectivity through VOIP hard phones. These soft exchanges shall be integrated with existing exchanges through PRI/ appropriate connectivity to make seamless communication within Assets/ Basins.

5.1.1 The VoIP Exchange:

a) The VoIP Exchanges shall be designed as per the requirements indicated at Functional requirements and Technical specifications. The system
shall provide Unified Communication solution at Delhi and VoIP Exchanges at Asset Headquarters.

b) The VoIP Exchange shall be in redundant configuration shall be installed at all Asset/Basin Headquarters as per BOM. All remote SS of this network shall extend voice connectivity through hard IP phones, registered with this IP Exchange. The system shall have IP architecture, and SIP enabled. It shall provide support for integrated telephony solution using IP phones and Media gateways allowing connectivity to analog as well as digital trunks, other H.323 and SIP enabled systems and video telephony.

c) The system should be able to scale up to numbers of IP phones as specified in Technical Specifications Part-IVB against each Asset/ Basin and shall support all current voice signaling standards/ protocols. It shall support efficient voice coding mechanisms and shall be able to support voice connectivity to IP phones behind NAT. The operating system should be hardened and updated with latest patches.

d) VoIP exchange Server shall have PRI ports for integrations with other existing exchanges.

e) Licenses for users as specified in Technical Specifications Part-IVB against each Asset/ Basin for each Exchange or as specified elsewhere in the bid document.

f) Hard VOIP phones shall be compatible with proposed VOIP Exchanges. Integrations of existing exchanges through E&M/ PRI trunks by using VOIP gateways and VOIP gateways shall have evolved appropriate voice compression technique.

g) Exchanges shall have 20% expandability provision form the number of ports specified.

h) Exchanges shall be capable of processing multiple calls simultaneously and maximum as per licenses.
5.1.2 Integration with existing legacy exchanges:

a) Various Asset/Basin head quarters and offices are having dedicated legacy PBXs. The make of these PBXs is given in the functional requirement details of each site. The subscribers of these exchanges should be able to seamlessly call remote subscribers of the central IP PBX, connected over Radios and vice versa. The design and implementation of dial plan should be such that it is in line with the existing dial plans of the legacy exchanges. The dial plans for integration with all existing legacy exchanges present in the network, should be centrally manageable.

b) In all cases, whether the call is between subscribers of the central IP PBX or between subscriber of IP PBX and subscriber of any legacy exchange in the network, the media streaming of real time traffic should be directly between the calling site and the called site over single satellite hop.

c) The proposed VoIP solution should be based on the best in the industry products. The system must be highly reliable with an availability of 99.99%.

d) The system shall include Media servers/ Gateways etc., for providing the comprehensive VoIP solution.

5.1.3 Media Server (Soft Exchange):

The Media Server shall provide centralized, class call processing.

The Media Servers shall support:

a) Distributed IP Networking and centralized call processing across multi-service networks

b) Dual server design with hot fail-over

c) Redundant LAN Interfaces and remote survivable call processing.
d) It shall include Communication Manager. The Communication Manager shall provide system management functionality, intelligent call routing, application integration and extensibility, and enterprise communications networking.

e) The Media server shall be configured in 1+1 hot standby mode.

5.1.4. Gateways:

Gateway shall support both bearer traffic and signaling traffic that is routed between packet-switched networks and circuit-switched networks. The Media Gateways shall contain the network and the endpoint interfaces, as well as call classification, announcement boards, and soon.

5.1.5. Management Functions:

a) The System shall have integrated management facility for managing converged voice and data networks. The applications include network management, fault management, performance management, configuration management, directory management, and policy management functionality.

b) Media Server/Gateway shall have PRI ports for integrations with other existing exchanges.

c) Soft exchange shall be complaint to H.323 and/or SIP and support PRI, DID, CO, FXO and FXS.

d) Licenses for number of users as specified elsewhere in the bid document.

e) Hard VOIP phones shall be compatible with proposed Soft Exchanges.

f) The system shall have appropriate voice compression capability for effective utilization of available bandwidth.

g) Integrations of existing exchanges through E&M trunks by using VOIP gateways. VOIP gateways shall have evolved appropriate voice compression technique complaint to G723 & G729 standard, support H.323 protocols, QSIG, echo cancellation.
h) The Voice Quality shall have better than 3.6 Mean Opinion Score (MOS).

i) Shall have LAN port 10/100 base T, Management SNMP, FXO/ FXS and 4wire E&M ports.

j) Soft Exchanges shall have 20% expandability provision form the number of ports specified.

k) Soft exchanges should be capable of processing multiple calls simultaneously and maximum as per licenses.

5.1.6. Media Server (Soft Exchange) capacity and number of VOIP extension shall be as follows:

a) Assam Asset: Equipped with 500 User licenses Expandable to 2000 IP users

b) A&AA Basin, Jorhat: Equipped with 500 User licenses Expandable to 2000 IP users

c) Tripura Asset: Equipped with 100 User licenses Expandable to 300 IP users

d) CBM Project: Equipped with 100 User licenses Expandable to 300 IP users

e) Rajahmundry Asset: Equipped with 500 User licenses Expandable to 2000 IP users

f) Cauvery Asset: Equipped with 500 User licenses Expandable to 2000 IP users

g) SCOPE Minar, Delhi: Equipped with 500 user licenses, expandable to 3000 users and meeting UC functionalities.
5.2. DESIGN CRITERIA FOR VOIP TELEPHONES INSTRUMENTS

a) VOIP telephone instrument shall be compatible, with the Soft Exchanges.

b) VOIP telephone instrument shall have hands-free dialing and Push button dialing.

c) VOIP telephone instrument shall have duplex speakerphone and echo cancellation.

d) VOIP telephone instrument shall have two ports and shall support DHCP.

e) VOIP telephone instrument shall have minimum 2X24 character LCD display.

f) VOIP hard phone shall be complaint to IEEE 802.3af standard, and shall be equipped with required Power adaptor.

5.2.1 DESIGN CRITERIA FOR EXTENDING VOIP CONNECTION /DATA

a) The cable used for extending voice and data connectivity shall have complete shielding.

b) UTP cables for extension of hard VOIP telephone up to users and distance shall be restricted to distance driven capability of data through UTP cables.

c) UTP cables for extension of data up to users, servers & switches and distance shall be restricted to distance driven capability of data through UTP cables.

d) UTP cables shall have low attenuation and complete shielding.

e) UTP cables shall withstand rigorous temperatures of 0 to 55°C and harsh humid/ marine environment.

f) UTP cables shall be FRLS [Flame Retardant Low Smoke]type.
g) For Users beyond 90m from Equipment: OFC with suitable Media Converter for extension of hard VOIP telephone and data.

h) OFC shall be Multimode & minimum diameter shall be as per Technical specifications.

i) OFC shall have low attenuation and shall be armoured for outdoor

j) OFC shall withstand rigorous temperatures of 0 to 55°C and harsh marine environment.

k) OFC cables shall be FRLS [Flame Retardant Low Smoke] type.

l) Fire Retardant Cables shall comply with the requirements of IEC 60332-3 (Category – C).