THE GAUHATI HIGH COURT

(High Courts of Assam, Nagaland, Mizoram and Arunachal Pradesh)

NOTICE INVITING QUOTATION

Dated 03/02/2021

Sealed quotations affixing Court fee stamps of Rs. 8.25 (Non-refundable) are hereby invited from interested firms for supply of an updated EPABX machine to the Gauhati High Court along with buyback of the two old machines. The last date for submission of the same is 22.02.2021. The firm should have at least 5 years of experience in relevant fields in Government offices. The details and specifications of the EPABX machine can be seen at Annexure-1. The firms are requested to submit the technical and financial bids separately.

Interested bidders are requested to make a survey of the High Court Registry before submitting their final bids. Any additional information may be obtained from the undersigned during office hours.

By Order
Sd/-
Joint Registrar (Finance)

Memo No HC.VIII-15/2010/ 2316 /AC Dated 03/02/2021

Copy to:-

1) The Directorate of Information & Public Relation, Dispur, Guwahati-06 (Janasanyog). He is requested to publish the Notice inviting Tender (NIT) urgently in two widely circulated local dailies(one English and one Assamese) and intimate this Registry regarding publication of the same.
2) The Project Manager/ System Analyst/ Programmer, Gauhati High Court, Guwahati. He is requested to upload the same in the High Court website.
3) High Court Notice Board.

5/2/21

Joint Registrar (Finance)

3/2/21

ANNEXURE - 1

EPABX Specifications

I. System Features:

- The offered system should be the latest make and model.
- The system should be state of the art and deployable over both packet and circuit switching infrastructure. The packet and circuit switching infrastructure shall be attainable through an interface board in the chassis itself. Detail architecture should be explained by the prospective bidders.
- The system should provide advanced Server Gateway solutions. It should have strong convergence solution for voice, LAN , Multiple VLAN, WAN Port, Inbuilt router ,Inbuilt Firewall, VPN, Voicemail, CDR, Call recording, Auto attending, IVRS without the need for any additional servers.
- The system should support traditional telephony, VOIP features and mobile telephone features in one single system.
- The system should be fully modular, scalable to minimum 512 ports. There should not be any change of the CPU or the system software during expansions
- The system shall run over an open source operating systems (Ubuntu, CentOS, etc) with native internet protocols
- The system should be modular and slandered at every level. It should be rackable, stackable.
- The system should have built-in SIP TRUNK Feature.
- The system should support telephony TCP/IP, UDP, ARP, ICMP, TELNET, SNMP, HTTP, RTP/RTCP, TFTP, NAT, DHCP, PPPoE, etc.
- The system should provide a single management interface window for all application management. Web browser based management software should be provided for the entire network management.
- System should be offered with inbuilt capability to support at least 96 SIP trunks to interlink with various other exchanges.
- System should be offered with inbuilt personal assistant.
- System should be offered voicemail for all extension.
- System should support seem less transfer of calls from desktop, mobile and laptops.
- OEM should have IP phone of same make.
- The media gateways that are geographically separated should have two 10/100/1000/mbps LAN interface with half/full duplex in auto negotiation.

28|1|21

II. Voice features:

2.1 The system should support the following voice terminals:

- Analog telephones
- Range of wireless telephones
- Range of IP hard phones
- High-end IP soft phones
- Client / server based solutions

2.2 It should support the following telephone features:

- Music on hold
- Personal assistant
- Voice mail
- Call forwarding
- Call transfer
- 3-way conference
- Directory
- Office greeting
- Hunting group (cyclical, sequential, parallel, IVR)
- Pick up (group, individual, supervised line)
- Boss / Secretary profile
- Automatic Route Selection (ARS)
- LDAP Directory structure.
- Self-number checking

2.3 The system should have rich set of telephony features

IP phones should provide minimum 6 SIP Accounts,2.4" 320X240 Colour Screen LCD, HD Voice, PoE Enabled, 10/100 Mbps Ethernet, Dual bridged port for PC Bypass, Handset/Hands-Free/Headphone Mode, Desktop/Wall – Mountable

2.4 Up to five different options should be offered to the incoming extension callers to reach different destinations like auto attendant, IVRS, secretary, mobile phone, outside number, operator, voicemail. This facility should be enabled for all the users in the network

2.5 The system should have in skin voicemail expandable to minimum 8 ports with minimum 200 hours of storage. The system should be offered with minimum 2 ports, 5 hours of recording.

2.6 The voice mail should be easy to use. User should be able to navigate through the voice mail features using voice prompts and soft keys of the IP phones. It should be possible to navigate and access all the voicemail features through the soft keys of the phone.

2.7 Following features should be supported:

- Direct access to any message irrespective of rank
- Record online function
- Screening function
- Notification by message LED
- Remote consultation to mailbox
- External notification (on mobile, home set, etc.)
- Personal options: customized greeting, protection by password, resend with comments, dial by name, reply function key
- Answer only mode
- Unconditional / on busy / on no answer forward on voice mail with specific message.

2.8 Teamwork features - Boss-Secretary features

28/1/21

- Boss call screening by secretary
- direct call between boss and secretary
- supervision of each other's set

2.9 Workgroup features for first day of installation

- with one single phone number for a group
- set monitoring
- call pickup
- broadcast – announce messages to the group's phone using loudspeakers
- Unanswered call notification.

2.10 Call screening

User should be able to listen to the messages as they are being left and choose the call, if the user wants to talk to the caller

2.11 Call recording

User should be able to record the conversations online for all extensions and all trunks.

- 3-way conference calls shall be possible.
- Inbuilt minimum 32 port IVRS
- Inbuilt Voice mail for all extension
- Inbuilt CDR

2.12 VOIP solutions

1. IP Telephony

   a. Just by plugging it into any Ethernet connection.
   b. Call server should support complete range of IP telephony devices, like -
      - a full range of integrated IP phones making Telephony over IP simple and transparent to the user
      - a soft phone which is installed in IP mode on a multimedia PC
      - A standard SIP set or any PC with the right SIP software (open source software).
      - Remote IP phones should be manageable by the call server at the central location.
      - It shall also support remote workers, through an embedded IP VPN service without any additional external hardware.
   c. Call server should have an embedded DHCP server, so that moving or adding an IP phone can happen

2. The system should support native VOIP solutions without any extra hardware. It should not involve any external gateway.

3. IP phones should have two Ethernet 10/100Base T ports, out of which one should be used to connect to LAN switch port and other one for connecting the desktop PC, thus saving LAN ports.

4. QoS features should be supported. It should be able to tag the voice packets based on standards using TOS / DiffServ etc.

5. The system should support the following compression algorithm for VOIP:
   - G711 when packets will stay in the LAN
   - G723.1 or G729.a when packets will be sent over the WAN

III. Networking:

1. Branch office and remote worker solutions

   It should be possible to have Remote IP phones in the branches, managed by the call server in the central location. The IP phones at the remote site should have exactly the same features as it is available at the central location.

2. The system should have networking features and it should allow to build the networks over the following physical interfaces:
   - ISDN
   - Leased lines
   - IP networks

3. The system should allow building future transparent networks. It should support the following features when networked:
   - Basic call (Internal/External outgoing call)
   - Block dialling
   - Call forwarding indication at operator console
   - Called party state indication on display
   - Caller line identification/caller line identification restriction
   - Diversion / Dynamic Routing
   - Dual Tone Multi Frequency transparency
   - Private / public call differentiation
   - Transfer

4. IP trunking
   - The call server shall support voice in the enterprise data WAN which shall include both voice and fax communications according to T38 standard.
   - The call server should ensure service transparency for the users.
   - In order to keep good voice quality, in case of problems in data network, the system shall overflow the new calls over ISDN

IV. System management and call accounting application:

GUI based software should be offered for the system management and call accounting. The software should have the following feature:

- Interactive user interface
- Access control by password
- Time charging (VPN cost simulated, tie line)
- Modifiable costs (over charging for re-invoicing)
- ISDN pulsing: Voice, Data at 64 Kb/s, complementary services invoiced upon use
- IP field taken into account in the ticket.
- Masking of the last digits of the dialled numbers
- Traffic analysis (incoming & outgoing calls)
- Total reports, hit-lists, cost thresholds
- Detailed and analysis report (crossed sorting)
- Account codes report
- Customized report
- Graphical presentation (diagrams, sectors, histograms)
- Recapitulative report (created from other reports)
- Automatic operations (report edition, data saving)
- Multi-carrier access
- Call back capability for secured access
- Any other related relevant features