

KSCC/SA/IT/5/7/2024

Date: 22/07/2024

ദർഘാസ് നോട്ടീസ്

ആലപ്പുഴ ഹെഡ് ഓഫീസിൽ നിലവിലുള്ള ടെലിഫോൺ ഇൻറ്റർകോം സേവനം IPBX -SIP TRUNK with VPN Router സിസ്റ്റത്തിലേക്ക് മാറ്റി സ്ഥാപിക്കുന്നതിലേക്കായി ദർഘാസുകൾ ക്ഷണിക്കുന്നു. ദർഘാസുകൾ 27/07/2024 ശനിയാഴ്ച ഉച്ചക്ക് 1 മണിക്ക് മുൻപായി ടി ഓഫീസിൽ ലഭിച്ചിരിക്കേണ്ടതാണ്. വിശദവിവരങ്ങൾക്ക് ടി ഓഫീസുമായി ബന്ധപ്പെടാവുന്നതാണ് .

PRE-QUALIFICATION CRITERIA

- The Bidder / Contractor should be a Manufacturer/Authorized Dealer / Sole Importer / Government Agency providing EPABX of latest proven design and should be established, reputed and experienced in the field of supply and maintenance of EPABX. OEM and quoted model existed in the Indian industry for last three years and proof of the same to be attached
- The vendor or OEM should have supplied, installed and commissioned successfully at least 2 IP based EPABX of proven design with at least 1000 Extension and 100 IP Phones or above. in public/private sector.



Work Scope

- Proposed Telephone Exchange system shall comprise of following components mentioned below:
 - IP Voice Extension : 60 Nos
 - Voice Mail for all users : 1 LOT
 - SIP Trunk License (BSNL) : 30 Ch
 - Audio Conference Bridge 32 Party : 1 Nos
 - Soft-ether Router : 1 Nos
 - Analog Trunk : 4 Nos
 - Voice Logger for all Users : 1 Lot

- IP-PBX System should support 60 ports.
- The IPABX shall be compatible with SIP TRUNK will be available With Bsnl. Requirement of SIP Trunk channels are mentioned above.
- Bidder has to design, supply, install and commission a new system meeting the above and the various features given under System specifications and System features.
- RACK and Cables and other accessories, if any, needed for connecting the exchange to Coir MDF and Rack shall be in the scope of the vendor.
- The vendor shall supply MDF for system side termination. system MDF shall be under the scope of the bidder and proper cable trays also to be fixed by bidder for routing the jumper wires. Original Krone make disconnection modules to be used in the MDF. Vendor has to provide MDF at media gateway end also.



REQUIREMENT

- The voice network architecture and call control function shall be SIP based.
- The IP telephony system should be a TEC certified system.
- The system must be capable of supporting Analog, IP Telephones, and SIP based video desk phones.
- The system should support SIP Client on smart phone.
- All SIP phones must support the standard SIP protocol.
- The offered system must support gateways for Analog Extensions.
- The IP telephony system must support unified communication (UC) server & gateways architecture for SIP, PRI , NGN and Analog trunks connectivity.
- The offered system should provide call details report.
- The offered system should support smart phone as extension with Wi-Fi client.
- The offered system should support 9 by 9 level of IVR System.
- The offered system should support web-based receptionist console.
- The receptionist should be able to intercept the calls in listen, whisper, barge-in mode.
- It shall be possible to make Video calls without adding any modules or licenses.
- The IP telephony system shall have SIP based architecture and provide support
- for integrated telephony solution for Analogue and IP phones, PSTN Gateways over IP architecture.



- The offered system should record all incoming and outgoing voice calls on same hardware and no external hardware should be used for voice recording.
- The offered system should support audio conference bridge with GUI for meet me and dial in and dial out conference.
- The offered system should additionally support Video Conference Software with Browser based Video Client, Screen Share, and Presentation for 5 users X 5 Conference room.
- The offered system should have the facility to integrate with Email, WhatsApp & SMS.
- The offered system should have the compatibility of optical fiber connectivity or GPON Connectivity
- The offered system should support call center applications, CTI, VoIP, Telephony over IP and IP trunk facilities, to address future requirements.
- The offered system should have an inbuilt LAN port, which can be connected to the LAN. It should be possible to access the system from any node of the LAN for maintenance purpose. The maintenance software should be browser based.
- The communication servers should support redundancy mode.
- All servers must be support in a cluster mode. If one cluster server fails, one of the other cluster servers in the network must be able to take the complete load of the



calls automatically (without any manual intervention). All servers should have same database.

- The offered system should support calling feature from our website to the PBX extension using WebRTC technology.
- The offered system should have chrome extension facilities for click to call, and thereby enabling to make calls from any browser-based application.
- The offered system should have missed call notification. A separate GUI should list out all missed calls, and remove the calls, once the callback has happened from the system.
- The offered system should include call billing module without any extra charges.
- Should support N+1 Redundancy Architecture as well as 1+1 redundancy Architecture.
- Should support Remote Survival Nodes.
- In case of failure of one server, the SIP Phones, SIP Gateways should register with second server automatically - the solution should support this functionality in future with addition of servers.
- It should be possible to take system translation as well as operating system backups on an external removable medium or backup over the LAN into an FTP server.




- It should be possible to add more sites and users without the need to change the software and existing configuration.
- System should support the following SIP RFCs:RFC 3261 (SIP: Session Initiation Protocol)
 - RFC 3262 (Reliability of Provisional Responses in Session Initiation Protocol)
 - RFC 3263 (Locating SIP Servers)
 - RFC 3264 (An Offer/Answer Model with Session Description Protocol (SDP))
 - RFC 3265 (Specific Event Notification)
 - RFC 2327 (SDP- Session Description Protocol)
 - RFC 1889 and 1890 (RTP/RTCP)
 - RFC 3515 (REFER)
 - RFC 2833 (DTMF over IP)
- The UC platform must have distributed architecture and centralized control for all the sites in the network.
- The system must support server gateway architecture with external appliance servers.
- The server should have AC power supply.
- Minimum 3 years' experience in work field



Required documents to be attached

- Work completion certificate minimum 3
- Excellence certificate if any (optional)


22/07/24
System Analyst



പകർപ്പ്: (i)കെ.സ്.സി.സി നോട്ടീസ് ബോർഡ്
(ii)നോട്ടീസ് ബോർഡ് കയർ ഫെഡ്
(iii)നോട്ടീസ് ബോർഡ് ഫോമിൽ