INDIAN INSTITUTE OF TECHNOLOGY GOA

At Goa Engineering College Campus
Farmagudi, Ponda, Goa 403401
E-mail: purchase@iitgoa.ac.in

GSTIN: 30AABAI1653D1ZF
PAN: AABAI1653D
TAN: BLRI08261B

Enquiry No: IITGOA/2019-20/013 **Date: 19/06/2019**

IIT Goa invites sealed quotations in two bid form for the supply and installation of below mentioned items.

Sl. No.	Description of Item	Quantity
1	Hybrid EPABX system with 50 IP Phones and perpetual smart phone licenses. **(Detailed specifications attached)**	01

Terms and conditions:

1. Quotation must be valid for at least 90 days.
2. The GSTIN should invariably be mentioned in your offer.
3. Kindly attach MAF from specified OEM with enquiry number. **This is MUST.**
4. Kindly attach compliance certificate as given below along with the technical quote. **This is MUST.**
5. Prices:
   I) For Import Supplies:
      a) It is mandatory to quote price in CIF/CIP Mumbai basis only with separate cost breakup of transportation up-to IIT GOA.
      b) In case of Multiple options of same product, bidders are requested to quote only one best option and not multiple options.
      c) All local taxes, customs duty and clearance charges will be borne by the Institute as applicable.
      d) Payment terms: 90% payment by letter of credit and balance 10% will be paid by wire transfer after satisfactory installation and commissioning.

   II) For Indigenous Supplies:
      a) In case of Multiple options of same product, bidders are requested to quote only one best option and not multiple options.

b) Payment terms: Within 30 days after the delivery and installation of the item at IIT GOA.

6. Delivery and installation should be made within 4 – 6 weeks of getting a confirmed order.
7. The suppliers shall provide the banking details along with their quote on their letterhead duly signed and stamped.
8. IIT Goa reserves the right to accept and/or reject any/all bids without assigning any reason.
9. Quotations shall be submitted in two parts;
    1) **Part – I (Technical)** should contain all the technical details and specification of the product. It should contain unpriced bid along with terms and conditions, compliance certificates, proprietary certificates (if applicable), any other certificates/details etc. This envelope should be marked as "Technical Bid"
    2) **Part -II (Financial)** The financial bid of the above item should be in a sealed envelope marked as "Financial Bid" and should contain financial terms and conditions.
10. For any clarification, you may kindly contact Mr. Raghavendra Y. K. (E-mail: sysad@iitgoa.ac.in) and Stores & Purchase Department (E-mail: purchase@iitgoa.ac.in) till 28/06/2019.
11. All sealed quotations labelled as "Quotation for Matrix EPABX Hybrid system" must reach to the Assistant Registrar (Stores & Purchase), IIT Goa, at Goa College of Engineering Campus, Farmagudi, Ponda, Goa by 17.00 Hrs on or before 10th July, 2019".

Sd/-

Asst. Registrar (S&P)

Detailed Specification

1.	**Communication System Architecture:**		
	• The 200 Port communication system should be configured for: **27 Analog Extension** **50 IP Extension** **1 Single Port PRI Card** **50 IP Phones as per specification**		
	• The communication system should be latest state of art new generation SIP based for converged IP telephony deployment. The communication system should have 32/64 Bit RISC processor.		
	• Communication System should be scalable, distributable and modular and the operating system shall be LINUX based. System shall employ IP at its core with IP switching technology and 100% non-blocking		
	• The communication system should have VOIP and Voice mail server at its core .i.e. VOIP and Voice mail server should not consume any slot in system.		
	• System shall be converged communication with ability to run TDM and IP on the same platform, it should support Analog, Digital, IP Telephones, mobile smart clients, PC UC clients & SIP based video desk phones.		
	• The communication System should be built on a universal slot architecture and modular in design to enable seamless growth, by adding the desired necessary cards as and when required. Any peripheral card can be inserted in any slot of the platform, whereby it is possible to increase or decrease the interface of the system as per the requirement in future as per mentioned maximum requirement.		
	• The architecture of the System shall be capable of seamless migration to its maximum capacity by simply adding peripherals cards in the same chassis without compromising function/features of the system. The architecture should be non-stackable eliminating individual power supply for each chassis		
	• The communication system should have distributed architecture using single box or multiple boxes. All the entities in the network shall be configurable from any location. It should be possible to provide Voice mail and UC features to all the users of the solution.		
	• The communication system shall have multiple port interfaces such as analog extension lines, digital key phone, IP Extension, C.O. Line, GSM/3G, E & M Line, PRI/E1 and VoIP. All interfaces shall be in the form of expansion cards and can be plugged into the universal slots of the system as and when require in the future. VoIP and Voice mail server should not consume any universal slot.		
2.	**Communication System Capabilities:**		
	• The communication system must build up high reliable software architecture running on Linux operating system.		
	• The offered communication system should provide communications solutions over IP, no restriction should evolve in terms of quality of service, reliability and security		
	• The communication system should support networking of two or more communication system over IP infrastructure		
	• The system must be suitable with adequate interfaces to provide control of communication process		
	• Support of different standard concerning the connection of voice terminals		
	• Provision of different solution to support availability of voice services		
	• Provide open interfaces and standard protocols for current and future applications		
	• Enable networking of system via TDM and IP infrastructure.		

	• Support of session initiation Protocol (SIP) to provide interface connection to ITSP and service providers		
	• Support 64 port Voice mail server, Voice mail server shall support features like call queuing, multi-language support, multiple MOH		
	• The system shall provide IP functionality at its core to support SIP/IP extensions and trunks over SIP protocol. It should be possible to support SIP Trunks and SIP/IP Extension with the single SIP server		
	• The system shall support in-skin voice mail server on CPU with 2170 hours of storage capacity and dedicated mailbox for each extension.		
	• It should have built-in multi-party conferencing. It should have minimum 15 conference possible of 3-party. The maximum number of participants required in single conference would be 20 or better.		
	• The system should have inbuilt Power failure transfer functionality on card. No external devices for Power failure required.		
	• The system shall have the inbuilt auto attendant facility and shall be able to answer minimum 9 or better calls simultaneously and should support dial-by-name.		
3.	**<u>Power supply:</u>**		
	• System power supply should be inbuilt and SMPS type, it shall also work on 230 V AC or 48 V DC + 20% to – 15 %.		
	• System should have LEDS on card to know power supply Health status		
4.	**<u>Interface Connectivity:</u>**		
4.1	**IP Trunk:**		
	• The system should support VoIP solutions as an integral part of the system. • The VoIP media gateway should not consume any universal slot. Universal slots should be usable for TDM port expansion. VoIP should be implemented by plug and play daughter board on server card • Support of minimum 90 SIP trunk from Day 1, SIP trunk should be License free. • System should have capability to support Video call Over IP • The system must support following features of IP telephony: Dynamic DNS, Registrar Server, Proxy Server, Presence Server, NAT and STUN, voice codec G.711u, G.711a, G.723, G.729, GSM, iLBC. • Only trusted IP address should be allowed for calling via Peer to Peer trunk. • Digest authentication shall be supported on peer to peer SIP trunk. • At least 500 IP addresses should be allowed in Trusted IP List.		
4.2	**CO Trunk :**		
	• 1200 ohm or better loop resistance • Loop start or ground start signaling • The CO trunk card so offered should extend CLI from DOT junction on to Analog Telephones, Digital Telephones and IP Telephones. • Should accept/extend CLIP features (FSK V.23, FSK Bell core 202 & DTMF) • Should support impedance matching. • Power failure transfer capability should be supported on card without addition of any external device • Support of AC impedance test for clear, audible and Echo free speech over CO trunk.		
4.3	**ISDN E1/PRI Trunk :**		

	• The system shall have an ISDN Digital platform and shall be compatible with ISDN PRI line of Local Service Provider, system should support networking over PRI with fiber optic connectivity. External media converter should not be required. • The offered exchange should be an ISDN ready switch. The system platform should always be ready for ISDN and only the necessary in skin ISDN cards need to be added for functionality • System should support one PRI port on single card with fiber and copper combination. • The offered system should be QSIG compliant (for PRI) for networking and Feature Transparency between two or more Systems. • System shall support bifurcation of E1 channels for voice and data. • The offered system shall support configurable E1 CAS Cards. • It should be possible to terminate fiber patch cord for connecting 2MB on any of the E1 ports. • Should support DTMF, MFCR2 and QSIG standard of interoperability between two exchanges		
4.3.1	**QSIG Features:**		
	• System Network support (Main exchange and Satellite exchanges) should support international telephone standard like QSIG for interoperability between two exchanges. • QSIG-BC-SS. • Heterogeneous, open numbering plan. • Calling/Connected Line Identification Presentation and Restriction. • Calling/Connected Name Identification Presentation and Restriction. • Call Forwarding Unconditional, Busy, No Reply, Call Transfer. • Call Completion to Busy Subscriber, on No Reply, call Offer. • System should have options to network over IP and ISDN technologies. • System must support the following external telephony interface signaling:-		
5	**<u>Subscriber Cards:</u>**		
5.1	**Analog Subscriber:**		
	• Each Port to support CLIP Feature both DTMF and FSK. The presentation of CLIP should be card dependent and should not be dependent on Central Resources. • Should be work on copper 0.5 mm cable without degradation of service up to minimum distance of 8 Kms. • The loop resistance of the subscriber card should be at least 1800(inclusive of the phone resistance) ohms or more.		
5.2	**Digital Subscriber:**		
	• Each Port to support CLIP feature for both DTMF and FSK. There should no compulsion of using different hardware for DTMF or FSK CLIP. All the ports should support both DTMF and FSK CLIP. • Should be work on copper 0.5 mm cable without degradation of service up to minimum distance of 500 meter or more. • Should have capacity to receive 10 or more calls simultaneously.		
5.3	**IP Subscriber :**		

	• The communication system should support all known 3rd party SIP phones • IP Subscriber should be able to register any IP hard Phone/soft phone over the entire network • Varied type of open SIP IP Terminals such as IP Phone, SIP soft phone and Mobile SIP Client shall be supported. • System shall also support an application for Android and iPhone so that the enterprise mobility can be extended for the Smartphone users. • Supplied IP Phones and PC based soft phones should be of the same manufacturer • IP phone operational functionality should be same as Digital Extension of PBX • System should have capability to support Video call Over IP		
6.	**<u>System Security :</u>**		
	• The system must incorporate advance security features like real time medial encryption. • System should have facility to disable Telnet and FTP server access to have maximum security. • Support of SIP over TLS and SRTP without any licenses. It should be IPv6 ready from first day • System SIP trunk must accept traffic from trusted IP source and it must support digest authentication for security of SIP traffic. • System should block GUI access after certain unsuccessful attempts of login. Also it should support Password Ageing. • SMS and Email notification should be sent of all Activity and Fault logs to predefined minimum two mobile numbers and two Email respectively.		

	• Operating System used by the communication system must not use or natively support network resource sharing services such as NFS, samba, LPR etc.
	• Password and access control must include at least:-
	  ▪ Shadow Passwords to prevent the possibility of an aggressor to easily read or deduce system or account access passwords.
	  ▪ Password Aging with configurable time periods.
	  ▪ Usage of MD5 algorithm (or stronger) for password encryption.
	• Internal OS controls for remote point of access restriction and service availability. (i.e. TCP Wrappers and Trusted Hosts)
	• IP Phones should not support direct, external initiated, connections via HTTP, telnet, FTP, TFTP or any other protocol as means to prevent distributed Denial of Service attack exploitation.
	• IP Phones must support 802.1x (EAP-MD5 or better) for authentication and access control to the network, this mechanism must allow the user to be connected to the system once he has passed the authentication process; not before
	• System should have options to configure voice VLAN number , allowing for the separation of voice and data traffic
	• Administration users connecting directly to the Call Server (console) shall be authenticated
	• All management traffic between a remote console/session and the system must be encrypted. (HTTPS for web sessions etc.)
	• The management platform must provide Role Based Account Management to define different levels of administrator access depending on specific function responsibility.		

7.	**<u>Software Up gradation ,Maintenance & Management:</u>**		
	• Web based GUI for maintenance, administration and configuration. Dedicated programing terminal should not be required. • System should support remote configuration Over WAN or any computer in LAN • System should support up gradation with direct GUI upload and there must minimum breakdown while up gradation of system. • The system can be programmed through Ethernet directly with online GUI without any external devices or modem. • The system shall have a built-in remote maintenance facility. The system can be programmed remotely over the internet without any modem required on the System side • System should maintain logs of all faults occurred. • Provision of notifying system admin by sending SMS on specific mobile number and by sending email on specific email Id.		

	• Reports of faults/activity should be available through GUI in Printable format • System should have buffer of storing minimum 500 faults/error logs • System Fault logs should be available in online/offline mode • Fault log notification on operator console. • SNMP support for warning messages traps, Errors must be sent to SNMP server with any SNMP v1, v2 & v3 protocols. • System should have notification of all alarms, logs to be saved in system. • System shall support notification of faults over SMS. • System must allow simultaneous GUI login from with 3 levels of access • Management platform must provide a single graphical client (Graphical User Interface (GUI)) • Management platform must provide web access allowing the administrator to manage the system to use any PC with an internet browser. • Configuration and Programming of services, users, categories and all system parameters and features. This must provide management in local or remote environments of a single system or a network. The network manager will be able to configure all locations users • Fault and error log notification availability on NMS software • Generate reports about faults and errors in PDF format • Access to exchange directory should be available through system GUI.it should be possible to Export/import Exchange directory in excel, csv format. • Integrated Directory. It shall be possible to provide display equipped voice terminals with access to system directory on digital and IP phones. • System shall have the provision of automatically identifying and isolating faulty trunks. This should be done on daily basis automatically or invoked by System administrator and results output on Maintenance terminal. • It should relieve the user from remembering lengthy procedure and formats for data changes and shall use simple English commands. • It should have options to record voice help message that can be available to all users by dialing voice help code. • The administration should be secured by way of password protection. It should be possible to have different layers of password. • The system management terminal shall be capable of adding/ modifying hardware/software at main location and remote Location from main location only • **System usage**: The usage display should enable users to view the results of running calls. • **Performance/status/information.** The software, through real time monitoring should be able to provide the status of extension and trunks to the EPABX performance at any time on request basis through screen displays • **Fault Detection/Alarm:** The software should constantly monitor the exchange performance and report/generate suitable alarms during any failures to indicate/localize the faults besides keeping the error-logs for various hardware failure detected.		
8.	**<u>Certification and Environmental Specification:</u>**		
	• The offered system shall be compatible with tropical climate prevalent in India • The system should be able to operate in relative humidity of 0-95% • System should be Fully operational between Temperature 0^oC to $+45^oC$ and storage temperature should be -20^oC to $+70^oC$ • Valid TEC certificate should be available for proposed model System should be certified for EMI/EMC, CE, RoHS and FCC15B.		
9.	**<u>System Features:</u>**		
	• **<u>Mobility:</u>** System shall support UC client application for Android/IOS and windows platform for extension mobility on smartphones and computers. It shall have features like Outlook Integration, BLF, Drag and Drop Conference and other Telephony Features.		
	• System shall support in-skin GSM card with Features like RCOC (Return Call to Original Caller), SIM Balance Check, BHCC selection, SMS notification on faults etc.		
	• System shall support SMS to Email and Email to SMS Feature without use of any additional software.		
	• System shall support SMS on NO reply.		

	• **<u>Call Routing:</u>** It shall support direct call routing of Trunk to Trunk call without intervention of Operator, Least cost routing, alternate number translation, Strip digit etc.		
	• Voice Message to be broadcasted can be recorded from telephone. It should not mandate recording the broadcast message on a PC.		
	• Voice Message can be broadcasted on mobile number of the users and also to the users of the networked systems.		
	• Support 16 different AC impedances on CO line to tune to any CO line for best speech performance.		
	• Last 250 unanswered calls made to external caller return back to original Caller.		
	• System forces the administrator to change password on first login and every 90 days for enhanced security of the system configuration.		
	• Call Back on Missed call.		
	• Bulk SMS up for sending emergency meeting invites		
	• **<u>Conferencing and built in Auto attendant:</u>** It shall support built in dial in and ad-hoc conference. Conferencing feature should be available from day 1 and it should be license free.		
	• It should have built-in multi-party conferencing. It should have minimum 15 conference possible of 3-party. The maximum number of participants required in single conference would be 21 minimum.		
	• Voice guided auto-attendant shall be preferably built-in.		
	• The system shall have the inbuilt auto attendant facility and shall be able to answer minimum 9 calls simultaneously and should support dial-by-name.		
	• **<u>Call billing:</u>** Detail reports of all system parameters should be generated through the CDR port of System. External third party Billing software shall not be required for basic report generation. Reports shall be directly saved in PDF format. Facility of online and scheduled report also should be available.		
	• **Call Accounting Data Export:** The Call Accounting Data Export feature should enable users to export information on station maintenance terminal. • Accounting of all calls generated by the users including cost, date, hour. Must provide different options to group the monitoring of the calls (extension number, trunk). EPABX system should have optimum storage buffer call details in case of Call billing system/ management system failed • **Import/Export Capability**. This should include easy graphical exporting and importing of directory numbers.		
	• The call ringing sequence would be programmable and have options such as simultaneous, hunting off, round robin and delayed simultaneous.		
	• System features shall have class of service, night service, conference, auto diagnostic etc. Class of service shall be unrestricted, STD restricted and semi restricted or customized.		
	• **Scheduling. Features**: The scheduling capability should enable users to specify a features to run at a specific time zone like call forward, schedule conference, scheduled message wait notification and scheduled call detailer report printing		
	• SNTP client should be inbuilt in System.		
	• The system shall have features as CLI based routing, call duration control, least cost routing i.e. time, number or combination of both.		
	• Extension features shall have an extension to extension call, extension to central office, extension to operator, automatic call back, call transfer, call forward, follow me, executive/secretary, do not disturb, barge-in, raid, Boss ring, Priority, emergency reporting etc.		
	• Operator features shall have the assistance to extension, attended call transfer, call intercept, indication of call waiting, night service control etc.		
	• The system shall have a built-in remote maintenance facility. The system can be programmed remotely over the internet without any modem required on the PBX side.		
	• The call ringing sequence would be programmable and have options such as simultaneous, hunting off, round robin and delayed simultaneous.		
	• Caller line identification (CLI) on Analog and digital/PRI trunks shall be built-in for both DTMF and FSK telephone instrument.		

	• Each port of the system shall be programmable. It shall have programmable features port-wise/extension-wise.		
	• The system shall support flexible numbering for extensions such as it may have extensions with 1 digit, 2 digits and up to 6 digits numbers as well as in combination of all.		
	• Access codes, system timers and access to features shall be programmable.		
	• Voice guided auto-attendant shall be preferably built-in.		
	• System should support dial form the directory. There shall be minimum 2900 numbers possible and shall also possible to dial it as an abbreviated number.		
	• Features given to an extension shall be accessed from any other extension by dialing the secret codes.		
	• System features shall have class of service, night service, conference, auto diagnostic etc. Class of service shall be unrestricted. STD restricted and semi restricted		
	• System shall support 128 VMS extensions over QSIG		
	• Assistance beep feature support while operator is on conversation,		
10.	**Other features:**		
	• Return call to original caller • 4 digit pin dialing, Virtual Stations • Emergency conference • Account Codes (Forced) • Allowed and Denied Lists, trunk reservation • Alternate Number Dialing • Backup CDR • Anonymous Call Rejection (SIP • Backup-System Configuration • Barge-in • Call Budget on Trunks • Call Cost Calculation • Call Duration Control • Call Progress Tones (Programmable, • Call Taping, Voice Prompts for Tones • CLI based Routing, Routing of calls to only permissible legal networks (Logical Partitioning) • Closed User Group (With/Without System ID) • Conference Dial-in • Conference – Multiple Participants • Conversation Recording • Daylight Saving Time (DST) • Direct Inward Dialing (DID) • Direct Dialing-In (DDI on T1/E1/PRI) • Direct Inward System Access (DISA) • Dynamic DNS (DDNS) • E-mail Notification (VMS) • Embedded Registrar and Proxy Servers (SIP Server) • Fax over IP (T .38 Relay and Pass-Through) • Hot Outward Dialing (With/Without Number & Delay) • Least Cost Routing (Number , Time and Service provider) • Live Call Screening (VMS) • Message Wait Indication • Multi-Stage Dialing • Online CDR • Peer-to-Peer Calling • Real Time Clock • Region Selection • CDR Posting (Call Accounting System Interface) • SMS Gateway & SMS server • Inbuilt Call Detail Records • System Activity Log and Display		
11	**IP Phone, Technical Specifications:**		
	Audio Features		

	• Full-duplex Hands-free Speakerphone with AEC • Codecs: G.711(A/µ), G.722, G.723, G.729, G.726, iLBC • DTMF: In-band, Out-of-band (RFC 2833) and SIP INFO • VAD, CNG, AEC, PLC, AJB, AGC		
	Phone Book		
	• Local Phonebook up to 1000 entries Black List • XML Remote Phonebook • Intelligent Search Method • Phonebook Search/Import/Export • Call History: Dialed/Received/Missed/Forwarded		
	Phone Features		
	• 1 VoIP Account • Call Hold, Mute, DND • One-touch Speed Dial, Hotline • Redial, Call Return, Auto Answer • Call Forward, Call Waiting, Call Transfer • Local 3-way Conferencing • Direct IP Call without SIP Proxy • Ringtone Selection/Import/Delete • Keypad Lock, Emergency Call • Set Date & Time Manually or Automatically • Dial Plan, XML Browser, Action URL/URI • Instant Messaging (Web UI and Phone)		
	Call Management		
	• Anonymous Call (CLIR) • Anonymous Call Rejection • Message Waiting Indicator (MWI) • Voicemail, Call Pickup • Intercom, Music on Hold • Call Completion, Hot-desking • Dial out Number from Web UI		
	Display		
	• 132x64-pixel Graphical LCD • LED for Incoming/Ongoing Call, MWI, Mute, Hold • Intuitive User Interface with Icons and Soft keys • Multiple Languages • Caller ID with Name, Number		
12	**Networking and Security**		
	• SIP v1 (RFC2543), v2 (RFC3261) • IPv6 • NAT Transverse: STUN Mode • Proxy Mode and Peer-to-Peer SIP Link Mode • IP Assignment: Static/DHCP/PPPoE • HTTP/HTTPS Web Server • Time and Date Synchronization using SNTP • UDP/TCP/DNS-SRV (RFC 3263) • QoS: 802.1p/Q Tagging (VLAN), Layer 3 ToS, DSCP • SRTP for Voice • Transport Layer Security (TLS) • HTTPS Certificate Manager • AES Encryption for Configuration File • Digest Authentication using MD5/MD5-sess • IEEE802.1X • SNMP v1/v2		
	Management		
	• Configuration: Browser/Phone/Auto Provision		

	• Auto Provision via FTP/TFTP/HTTP/HTTPS for Mass Deploy • Server Redundancy • Factory Reset • Soft Reboot • Package Tracing • System Log		
	Physical Features		
	• 2 x 10/100 Mbps LAN & PC Ports • 29 keys including 4 Soft Keys • 1 x RJ9 Handset Port • 1 x RJ9 Headset Port • Dimension (W x D x H): 185 x 188 x 143 mm		
	Power Supply		
	• Power Adapter: 5VDC/600mA (Included in the Box) • Power over Ethernet (IEEE 802.3af) • Power Consumption: 5W (Typical) • Connector: DC Power Jack		
	Mechanical		
	• Installation: Wall Mount, Table-top		
	Environmental		
	• Operating Temperature: -10° C to 50°C (14° F to 122° F) • Operating Humidity: 10 - 95% (Non-Condensing)		
	Certifications		
	• CE, FCC-15 (Class-B), RoHS		

Compliance Certificate

Name of the Firm : ____

Email Id :____

Mobile No :____

1.	**Communication System Architecture:**	**compliance**	**Remarks**
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	• The architecture of the System shall be capable of seamless migration to its maximum capacity by simply adding peripherals cards in the same chassis without compromising function/features of the system. The architecture should be non-stackable eliminating individual power supply for each chassis		
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	• The communication system shall have multiple port interfaces such as analog extension lines, digital key		

	phone, IP Extension, C.O. Line, GSM/3G, E & M Line, PRI/E1 and VoIP. All interfaces shall be in the form of expansion cards and can be plugged into the universal slots of the system as and when require in the future.VoIP and Voice mail server should not consume any universal slot.		
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	• The system shall support in-skin voice mail server on CPU with 2170 hours of storage capacity and dedicated mailbox for each extension.		
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3.	**<u>Power supply:</u>**	**<u>compliance</u>**	**<u>Remarks</u>**
	• System power supply should be inbuilt and SMPS type, it shall also work on 230 V AC or 48 V DC + 20% to – 15 %.		
	• System should have LEDS on card to know power supply Health status		
4.	**<u>Interface Connectivity:</u>**	**<u>compliance</u>**	**<u>Remarks</u>**
4.1	**IP Trunk:**		

	• The system should support VoIP solutions as an integral part of the system. • The VoIP media gateway should not consume any universal slot. Universal slots should be usable for TDM port expansion. VoIP should be implemented by plug and play daughter board on server card • Support of minimum 90 SIP trunk from Day 1, SIP trunk should be License free. • System should have capability to support Video call Over IP • The system must support following features of IP telephony: Dynamic DNS, Registrar Server, Proxy Server, Presence Server, NAT and STUN, voice codec G.711u, G.711a, G.723, G.729, GSM, iLBC. • Only trusted IP address should be allowed for calling via Peer to Peer trunk. • Digest authentication shall be supported on peer to peer SIP trunk. • At least 500 IP addresses should be allowed in Trusted IP List.		
4.2	**CO Trunk :**		
	• 1200 ohm or better loop resistance • Loop start or ground start signaling • The CO trunk card so offered should extend CLI from DOT junction on to Analog Telephones, Digital Telephones and IP Telephones. • Should accept/extend CLIP features (FSK V.23, FSK Bell core 202 & DTMF) • Should support impedance matching. • Power failure transfer capability should be supported on card without addition of any external device • Support of AC impedance test for clear, audible and Echo free speech over CO trunk.		
4.3	**ISDN E1/PRI Trunk :**	**compliance**	**Remarks**

	• The system shall have an ISDN Digital platform and shall be compatible with ISDN PRI line of Local Service Provider, system should support networking over PRI with fiber optic connectivity. External media converter should not be required. • The offered exchange should be an ISDN ready switch. The system platform should always be ready for ISDN and only the necessary in skin ISDN cards need to be added for functionality • System should support one PRI port on single card with fiber and copper combination. • The offered system should be QSIG compliant (for PRI) for networking and Feature Transparency between two or more Systems. • System shall support bifurcation of E1 channels for voice and data. • The offered system shall support configurable E1 CAS Cards. • It should be possible to terminate fiber patch cord for connecting 2MB on any of the E1 ports. • Should support DTMF, MFCR2 and QSIG standard of interoperability between two exchanges		
4.3.1	**QSIG Features:**	**compliance**	**Remarks**
	• System Network support (Main exchange and Satellite exchanges) should support international telephone standard like QSIG for interoperability between two exchanges. • QSIG-BC-SS. • Heterogeneous, open numbering plan. • Calling/Connected Line Identification Presentation and Restriction. • Calling/Connected Name Identification Presentation and Restriction. • Call Forwarding Unconditional, Busy, No Reply, Call Transfer. • Call Completion to Busy Subscriber, on No Reply, call Offer. • System should have options to network over IP and ISDN technologies. • System must support the following external telephony interface signaling:-		
5	**Subscriber Cards:**	**compliance**	**Remarks**
5.1	**Analog Subscriber:**		

	• Each Port to support CLIP Feature both DTMF and FSK. The presentation of CLIP should be card dependent and should not be dependent on Central Resources. • Should be work on copper 0.5 mm cable without degradation of service up to minimum distance of 8 Kms. • The loop resistance of the subscriber card should be at least 1800(inclusive of the phone resistance) ohms or more.		
5.2	**Digital Subscriber:**		
	• Each Port to support CLIP feature for both DTMF and FSK. There should no compulsion of using different hardware for DTMF or FSK CLIP. All the ports should support both DTMF and FSK CLIP. • Should be work on copper 0.5 mm cable without degradation of service up to minimum distance of 500 meter or more. • Should have capacity to receive 10 or more calls simultaneously.		
5.3	**IP Subscriber :**		
	• The communication system should support all known 3rd party SIP phones • IP Subscriber should be able to register any IP hard Phone/soft phone over the entire network • Varied type of open SIP IP Terminals such as IP Phone, SIP soft phone and Mobile SIP Client shall be supported. • System shall also support an application for Android and iPhone so that the enterprise mobility can be extended for the Smartphone users. • Supplied IP Phones and PC based soft phones should be of the same manufacturer • IP phone operational functionality should be same as Digital Extension of PBX • System should have capability to support Video call Over IP		
6.	**System Security :**	**compliance**	**Remarks**
	• The system must incorporate advance security features like real time medial encryption. • System should have facility to disable Telnet and FTP server access to have maximum security. • Support of SIP over TLS and SRTP without any licenses. It should be IPv6 ready from first day • System SIP trunk must accept traffic from trusted IP source and it must support digest authentication for security of SIP traffic. • System should block GUI access after certain unsuccessful attempts of login. Also it should support Password Ageing. • SMS and Email notification should be sent of all Activity and Fault logs to predefined minimum two mobile numbers and two Email respectively.		
	• Operating System used by the communication system		

	must not use or natively support network resource sharing services such as NFS, samba, LPR etc. • Password and access control must include at least:- ▪ Shadow Passwords to prevent the possibility of an aggressor to easily read or deduce system or account access passwords. ▪ Password Aging with configurable time periods. ▪ Usage of MD5 algorithm (or stronger) for password encryption. • Internal OS controls for remote point of access restriction and service availability. (i.e. TCP Wrappers and Trusted Hosts) • IP Phones should not support direct, external initiated, connections via HTTP, telnet, FTP, TFTP or any other protocol as means to prevent distributed Denial of Service attack exploitation. • IP Phones must support 802.1x (EAP-MD5 or better) for authentication and access control to the network, this mechanism must allow the user to be connected to the system once he has passed the authentication process; not before • System should have options to configure voice VLAN number , allowing for the separation of voice and data traffic • Administration users connecting directly to the Call Server (console) shall be authenticated • All management traffic between a remote console/session and the system must be encrypted. (HTTPS for web sessions etc.) • The management platform must provide Role Based Account Management to define different levels of administrator access depending on specific function responsibility.		
7.	**<u>Software Up gradation ,Maintenance & Management:</u>**	**<u>compliance</u>**	**<u>Remarks</u>**
	• Web based GUI for maintenance, administration and configuration. Dedicated programing terminal should not be required. • System should support remote configuration Over WAN or any computer in LAN • System should support up gradation with direct GUI upload and there must minimum breakdown while up gradation of system. • The system can be programmed through Ethernet directly with online GUI without any external devices or modem. • The system shall have a built-in remote maintenance facility. The system can be programmed remotely over the internet without any modem required on the System side • System should maintain logs of all faults occurred. • Provision of notifying system admin by sending SMS on specific mobile number and by sending email on specific email Id. • Reports of faults/activity should be available through GUI in Printable format • System should have buffer of storing minimum 500		

	faults/error logs • System Fault logs should be available in online/offline mode • Fault log notification on operator console. • SNMP support for warning messages traps, Errors must be sent to SNMP server with any SNMP v1, v2 & v3 protocols. • System should have notification of all alarms, logs to be saved in system. • System shall support notification of faults over SMS. • System must allow simultaneous GUI login from with 3 levels of access • Management platform must provide a single graphical client (Graphical User Interface (GUI)) • Management platform must provide web access allowing the administrator to manage the system to use any PC with an internet browser. • Configuration and Programming of services, users, categories and all system parameters and features. This must provide management in local or remote environments of a single system or a network. The network manager will be able to configure all locations users • Fault and error log notification availability on NMS software • Generate reports about faults and errors in PDF format • Access to exchange directory should be available through system GUI.it should be possible to Export/import Exchange directory in excel, csv format. • Integrated Directory. It shall be possible to provide display equipped voice terminals with access to system directory on digital and IP phones. • System shall have the provision of automatically identifying and isolating faulty trunks. This should be done on daily basis automatically or invoked by System administrator and results output on Maintenance terminal. • It should relieve the user from remembering lengthy procedure and formats for data changes and shall use simple English commands. • It should have options to record voice help message that can be available to all users by dialing voice help code. • The administration should be secured by way of password protection. It should be possible to have different layers of password. • The system management terminal shall be capable of adding/ modifying hardware/software at main location and remote Location from main location only • **System usage**: The usage display should enable users to view the results of running calls. • **Performance/status/information.** The software, through real time monitoring should be able to provide the status of extension and trunks to the EPABX performance at any time on request basis through screen displays		

	• **Fault Detection/Alarm:** The software should constantly monitor the exchange performance and report/generate suitable alarms during any failures to indicate/localize the faults besides keeping the error-logs for various hardware failure detected.		
8.	**Certification and Environmental Specification:**	**compliance**	**Remarks**
	• The offered system shall be compatible with tropical climate prevalent in India • The system should be able to operate in relative humidity of 0-95% • System should be Fully operational between Temperature $0^{o}C$ to $+45^{o}C$ and storage temperature should be $-20^{o}C$ to $+70^{o}C$ • Valid TEC certificate should be available for proposed model System should be certified for EMI/EMC, CE, RoHS and FCC15B.		
9.	**System Features:**	**compliance**	**Remarks**
	• **Mobility:**System shall support UC client application for Android/IOS and windows platform for extension mobility on smartphones and computers. It shall have features like Outlook Integration, BLF, Drag and Drop Conference and other Telephony Features.		
	• System shall supportin-skin GSM card with Features like RCOC (Return Call to Original Caller), SIM Balance Check, BHCC selection, SMS notification on faults etc.		
	• System shall support SMS to Email and Email to SMS Feature without use of any additional software.		
	• System shall support SMS on NO reply.		
	• **Call Routing:**It shall supportdirect call routing of Trunk to Trunk call without intervention of Operator, Least cost routing, alternate number translation, Strip digit etc.		
	• Voice Message to be broadcasted can be recorded from telephone. It should not mandate recording the broadcast message on a PC.		
	• Voice Message can be broadcasted on mobile number of the users and also to the users of the networked systems.		
	• Support 16 different AC impedances on CO line to tune to any CO line for best speech performance.		
	• Last 250 unanswered calls made to external caller return back to original Caller.		
	• System forces the administrator to change password on first login and every 90 days for enhanced security of the system configuration.		
	• Call Back on Missed call.		

	• Bulk SMS up for sending emergency meeting invites		
	• **Conferencing and built in Auto attendant:** It shall support built in dial in and ad-hoc conference. Conferencing feature should be available from day 1 and it should be license free.		
	• It should have built-in multi-party conferencing. It should have minimum 15 conference possible of 3-party. The maximum number of participants required in single conference would be 21 minimum.		
	• Voice guided auto-attendant shall be preferably built-in.		
	• The system shall have the inbuilt auto attendant facility and shall be able to answer minimum 9 calls simultaneously and should support dial-by-name.		
	• **Call billing:** Detail reports of all system parameters should be generated through the CDR port of System. External third party Billing software shall not be required for basic report generation. Reports shall be directly saved in PDF format. Facility of online and scheduled report also should be available.		
	• **Call Accounting Data Export:** The Call Accounting Data Export feature should enable users to export information on station maintenance terminal. • Accounting of all calls generated by the users including cost, date, hour. Must provide different options to group the monitoring of the calls (extension number, trunk). EPABX system should have optimum storage buffer call details in case of Call billing system/ management system failed • **Import/Export Capability**. This should include easy graphical exporting and importing of directory numbers.		
	• The call ringing sequence would be programmable and have options such as simultaneous, hunting off, round robin and delayed simultaneous.		
	• System features shall have class of service, night service, conference, auto diagnostic etc. Class of service shall be unrestricted, STD restricted and semi restricted or customized.		
	• **Scheduling. Features**: The scheduling capability should enable users to specify a features to run at a specific time zone like call forward, schedule conference, scheduled message wait notification and scheduled call detailer report printing		
	• SNTP client should be inbuilt in System.		
	• The system shall have features as CLI based routing, call duration control, least cost routing i.e. time, number or combination of both.		
	• Extension features shall have an extension to extension call, extension to central office, extension to operator, automatic call back, call transfer, call forward, follow me, executive/secretary, do not disturb, barge-in, raid, Boss ring, Priority, emergency reporting etc.		

	• Operator features shall have the assistance to extension, attended call transfer, call intercept, indication of call waiting, night service control etc.		
	• The system shall have a built-in remote maintenance facility. The system can be programmed remotely over the internet without any modem required on the PBX side.		
	• The call ringing sequence would be programmable and have options such as simultaneous, hunting off, round robin and delayed simultaneous.		
	• Caller line identification (CLI) on Analog and digital/PRI trunks shall be built-in for both DTMF and FSK telephone instrument.		
	• Each port of the system shall be programmable. It shall have programmable features port-wise/extension-wise.		
	• The system shall support flexible numbering for extensions such as it may have extensions with 1 digit, 2 digits and up to 6 digits numbers as well as in combination of all.		
	• Access codes, system timers and access to features shall be programmable.		
	• Voice guided auto-attendant shall be preferably built-in.		
	• System should support dial form the directory. There shall be minimum 2900numbers possible and shall also possible to dial it as an abbreviated number.		
	• Features given to an extension shall be accessed from any other extension by dialing the secret codes.		
	• System features shall have class of service, night service, conference, auto diagnostic etc. Class of service shall be unrestricted. STD restricted and semi restricted		
	• System shall support 128 VMS extensions over QSIG		
	• Assistance beep feature support while operator is on conversation,		
10.	**Other features:**	**compliance**	**Remarks**
	• Return call to original caller • 4 digit pin dialing, Virtual Stations • Emergency conference • Account Codes (Forced) • Allowed and Denied Lists, trunk reservation • Alternate Number Dialing • Backup CDR • Anonymous Call Rejection (SIP • Backup-System Configuration • Barge-in • Call Budget on Trunks • Call Cost Calculation • Call Duration Control • Call Progress Tones (Programmable, • Call Taping, Voice Prompts for Tones • CLI based Routing, Routing of calls to only permissible legal networks (Logical Partitioning) • Closed User Group (With/Without System ID) • Conference Dial-in		

	• Conference – Multiple Participants • Conversation Recording • Daylight Saving Time (DST) • Direct Inward Dialing (DID) • Direct Dialing-In (DDI on T1/E1/PRI) • Direct Inward System Access (DISA) • Dynamic DNS (DDNS) • E-mail Notification (VMS) • Embedded Registrar and Proxy Servers (SIP Server) • Fax over IP (T .38 Relay and Pass-Through) • Hot Outward Dialing (With/Without Number & Delay) • Least Cost Routing (Number , Time and Service provider) • Live Call Screening (VMS) • Message Wait Indication • Multi-Stage Dialing • Online CDR • Peer-to-Peer Calling • Real Time Clock • Region Selection • CDR Posting (Call Accounting System Interface) • SMS Gateway & SMS server • Inbuilt Call Detail Records • System Activity Log and Display		
11	**IP Phone, Technical Specifications:**	**compliance**	**Remarks**
	Audio Features		
	• Full-duplex Hands-free Speakerphone with AEC • Codecs: G.711(A/µ), G.722, G.723, G.729, G.726, iLBC • DTMF: In-band, Out-of-band (RFC 2833) and SIP INFO • VAD, CNG, AEC, PLC, AJB, AGC		
	Phone Book		
	• Local Phonebook up to 1000 entries Black List • XML Remote Phonebook • Intelligent Search Method • Phonebook Search/Import/Export • Call History: Dialed/Received/Missed/Forwarded		
	Phone Features		
	• 1 VoIP Account • Call Hold, Mute, DND • One-touch Speed Dial, Hotline • Redial, Call Return, Auto Answer • Call Forward, Call Waiting, Call Transfer • Local 3-way Conferencing • Direct IP Call without SIP Proxy • Ringtone Selection/Import/Delete • Keypad Lock, Emergency Call • Set Date & Time Manually or Automatically • Dial Plan, XML Browser, Action URL/URI • Instant Messaging (Web UI and Phone)		
	Call Management		
	• Anonymous Call (CLIR) • Anonymous Call Rejection • Message Waiting Indicator (MWI) • Voicemail, Call Pickup • Intercom, Music on Hold		

	• Call Completion, Hot-desking • Dial out Number from Web UI		
	Display		
	• 132x64-pixel Graphical LCD • LED for Incoming/Ongoing Call, MWI, Mute, Hold • Intuitive User Interface with Icons and Soft keys • Multiple Languages • Caller ID with Name, Number		
12	**Networking and Security**	**compliance**	**Remarks**
	• SIP v1 (RFC2543), v2 (RFC3261) • IPv6 • NAT Transverse: STUN Mode • Proxy Mode and Peer-to-Peer SIP Link Mode • IP Assignment: Static/DHCP/PPPoE • HTTP/HTTPS Web Server • Time and Date Synchronization using SNTP • UDP/TCP/DNS-SRV (RFC 3263) • QoS: 802.1p/Q Tagging (VLAN), Layer 3 ToS, DSCP • SRTP for Voice • Transport Layer Security (TLS) • HTTPS Certificate Manager • AES Encryption for Configuration File • Digest Authentication using MD5/MD5-sess • IEEE802.1X • SNMP v1/v2		
	Management		
	• Configuration: Browser/Phone/Auto Provision • Auto Provision via FTP/TFTP/HTTP/HTTPS for Mass Deploy • Server Redundancy • Factory Reset • Soft Reboot • Package Tracing • System Log		
	Physical Features		
	• 2 x 10/100 Mbps LAN & PC Ports • 29 keys including 4 Soft Keys • 1 x RJ9 Handset Port • 1 x RJ9 Headset Port • Dimension (W x D x H): 185 x 188 x 143 mm		
	Power Supply		
	• Power Adapter: 5VDC/600mA (Included in the Box) • Power over Ethernet (IEEE 802.3af) • Power Consumption: 5W (Typical) • Connector: DC Power Jack		
	Mechanical		
	• Installation: Wall Mount, Table-top		
	Environmental		
	• Operating Temperature: -10° C to 50°C (14° F to 122° F) • Operating Humidity: 10 - 95% (Non-Condensing)		
	Certifications		
	• CE, FCC-15 (Class-B), RoHS		