भारत सरकार

GOVERNMENT OF INDIA

परिवार कल्याण प्रशिक्षण तथा अनुसंधान केंद्र

FAMILY WELFARE TRAINING & RESEARCH CENTRE

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No. FW/EPABX System Purchase/2018-19/ Date: 22/02/2019

NOTICE INVITING TENDER

Quotation for Purchase of EPABX System at New Institutional & Hostel Building of FWTRC at Plot No 6 & 6A, Sector-18, New Panvel, Navi Mumbai.

On behalf of Director, FWTRC, Mumbai and Ministry of Health & Family Welfare, Govt. of India, sealed Quotations are invited from vendors who have sufficient experience in the field and have worked with Government Organisation for **Purchase of EPABX System of following Specification.**

Sr.No.	Nomenclature	Qty
	(a) System Architecture	
	Hybrid Communication system with IP at core Expandable upto 200 extensions, 8 PRI and 80 digital extensions, 60 CO lines, 90 SIP trunks,900 IP extension on the same platform without adding any cabinet and with single power supply. The System should be configured as per below requirement from first day.	1
	(b) Subscriber cards	
	Analog subscriber with CLI ports	120
	Digital subscriber port	18
	(c) Trunk Cards	
	CO line (CLI based) port	4
	T1 E1 ISDN PRI Port	1
	IP trunks and Channels	20
	(d) User End Equipment	
	Digital Operator console with 16 inbuilt DSS keys	12
	Push button CLI phones	72
	(e) MDF	
	200 Pair MDF with krone module	1
	(f) Power Arrangements	
	Power Supply: 100-240 VAC, 47-60Hz	
	(g) Installation and commissioning	

Qualification Criteria:

• Manufacturer should have government approved R&D unit
• The manufacturer should be ISO Certified Company
• OEM should have worked and executed similar projects for more than 10 Years
• The Product quoted should be TEC approved
• Case specific OEM authorization is must
• OEM should be MSME with valid Udyog Aadhaar.
• Bidder should submit MAF for participation as an Authorised dealer from OEM of the product.

Quotations should include rates along with all applicable taxes. Quotations in sealed cover superscripted "Tender for providing EPABX System to be submitted to Director, FWTRC, Mumbai.

Last date and time for submission of tender is 11th March 2019 till 17:00 hrs.

Sd-
DIRECTOR,
FWTRC, 332 S.V.P. ROAD, KHETWADI, MUMBAI- 04
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1.	**Communication System Architecture:**
	• 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.	**<u>Communication System Capabilities:</u>**
	• It should be possible to reach the maximum capacity of system up to 200+ extensions, 8 PRI and 80 digital extensions, 60 CO lines,90 SIP trunks,900 IP extension,15 radio ports & 30 E&M and 40 GSM trunk on the same platform without adding any cabinet and with single power supply.
	• 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.
	• The system memory should be based on Flash EPROM for higher reliability and faster rebooting minimum storage capacity of 256 MB ROM,2GB RAM & 900 MHZ clock speed

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 %.
	• Maximum power consumption should not be more than 250 Watts/hours
	• System should have LEDS on card to know power supply Health status

4.	**<u>Interface Connectivity:</u>**
4.1	**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.
4.2	**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 99 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.
4.3	**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.	**System Security :**
	• 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. • System must be provided adequate protection from possible virus, worm and Trojan infestation points such as internal e-mail servers and they must be updated every month. The protection software must be valid for a period of three years with updates pushed in every 15 days. • 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.	**Software Up gradation ,Maintenance & Management:**
	• 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 • 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 • 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:**
	• 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% • The system should support 19” rack mount, table top mount. • 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.	<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.
	• <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.
	• <u>**Call billing:**</u> Detail reports of all system parameters should be generated through the SMDR 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 system should have built-in outgoing Call Log buffer of 5000 calls, incoming call log buffer of 5000 and call log buffer of 2000 internal calls.
	• 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.
	• System features shall have class of service, night service, conference, auto diagnostic etc. Class of service shall be unrestricted. STD restricted and semi restricted

10.	**Specification of Operator Console:**
	• Graphical LCD with Backlit • LED for Incoming/Ongoing Call, Mute, Hold • 16 memory/speed dial keys • Should be able to use server integrated directory • backlight display panel • LED for Incoming/Ongoing Call, Mute, Hold • Add on 32 key module support, maximum 4 key modules shall be supported • Should be possible to connect additional key modules up to 128 keys • Should be able to attach a headset to the operator console • Intuitive User Interface with Icons • Multiple Languages Caller ID with Name, Number • 45 or more keys including 4 Context Sensitive Hard Keys • 1 x RJ9 Handset Port, 1 x RJ9 Headset Port • 1 x 3.5 mm Headset Port • Installation: Wall Mount, Table-top, • CE, FCC-15. RoHS certified • Mute, Call Hold, Do Not Disturb, Speed Dial, Hotline • Redial, Call Back, Auto Answer, Call Forward, Call Waiting, Call Transfer • Room Monitoring, Conference, Directory, Call Logs ,Dial-by-Name • Message wait Lamp, Ringer Lamp. Voice Mail ,Call Pickup – Group and Selective

11	**Distribution Frames and Protection Modules.: MDF/IDF**

11.1	• The MDF shall be of Krone only, complete with all its accessories for interconnecting and jumpering the various terminations to and from EPABX.
	• Protection against over voltage above 95 volts and over current above 100 mA should be provided using self-healing protection devices. The protection devices should be of minimum 3 stage
	• The MDF shall be so designed that it is easy to work with.
	• The MDF shall mainly consist of the self-supporting frame work to mount the connector box, either on the wall or on the floor with sufficient no of jumper guides and rings.
	• Protector module shall be provided for all the terminated ports on the line side, whether they are extended to the equipment or not. The following accessories shall be supplied: -
	    • Dust cover (Or a closed door on cabinet)
	    • Number tag
	    • Punching Kit
	    • Any other accessory as deemed necessary by the bidder may also be quoted
	    • Additional protection for over voltage and undercurrent if any which is suggested may be quoted as an optional item. However, after procurement of these items the entire responsibility towards protection of EPABX against these eventualities will be that of vendor