



Ref. No. *ECO/TENDER/NOTICE/172/19-20*

Date: -03.02.2020

### Notice of Request for Proposal

Quotations for the following items are invited from authorized parties for SITCT the following items. Suppliers are hereby requested to submit current trade license, GST, PAN, MAF, TEC and other credentials within 20.02.2020 in a sealed envelope addressed to the undersigned. The authority reserves the right to accept/reject any quotation without any reason thereof.

Sl. No.	Items	Qty.	Make
01	Supply Installation Testing Commissioning Training of Embedded Server Based TEC Certified IP PBX System with 2 P&T 2 Digital Extensions 28 Analog Extensions along with 1 Digital Operator Console having 16 Built in DSS Keys, expandable upto 48 Analog & 50 IP Ports.	01	MATRIX/ALCATEL/ AVAYA/TADIRAN
02	Supply & Laying of 2 Pair PVC Telephone Cable along with ISI marked PVC Conduit.	500 M	FINOLEX/DELTON/ POLYCAB
03	Main Distribution Frame – 50 Pair along with ADC Krone Disconnection Module	01	ADC
04	Caller Line Identification based Analog Telephone Set	Unit Price	Beetel/Panasonic
05	Analog Telephone Set without Caller Line Identification	Unit Price	Beetel/Panasonic

\*\* Documents Need to be submitted along with quotation: -

1. OEM Authorization Certificate for this tender - for Sl. No. 1, 4 & 5
2. Technical Datasheet – for Sl. No. 1, 4 & 5
3. TEC Certificate – for Sl. No. 1
4. Compliance Sheet as per ANNEXURE A
5. Trade License, GST, PAN and Credentials of the bidder.

Head of the Department  
Department of Economics  
University of Calcutta

## ANNEXURE - A

Sr. No.	Requirement Specifications	Compliance Y/N	Remarks
1	The equipment should be electronic type. The system shall have microprocessor / micro controller based on Stored Program Control Technique. It should employ PCM/TDM, 100% non-blocking, digital switching technology.		
2	The system should have distributed processing architecture, SLIC and SMT Design.		
3	The 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 trunk lines or subscriber lines of the system as per the requirement in future as per mentioned maximum requirement.		
4	The architecture of the system should be capable of seamless migration to its maximum capacity by simply adding peripheral cards in the same chassis without compromising functions/features of the system. The architecture should be non-stackable eliminating individual power supply for each chassis.		
5	The system should have multiple port interfaces such as analog extension, digital key phone, IP extension, CO line, GSM/3G, 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 required in the future.		
6	The system should have combo cards (any combination of PSTN, DIGITAL and ANALOG) to have flexible configuration and save on the resources of universal slots.		
7	The system should retain traditional networks (CO, ISDN) along with access to VoIP and GSM networks in single platform just by adding expansion cards		
8	The system should be suitable for DTMF as well as the FSK type of telephone instruments.		
9	The system should be possible to reach the capacity of main system up to 48 extensions, 6 PRI and 32 digital extensions, 16 CO lines on the same platform without adding any cabinet and with single power supply.		
10	The system power supply should be inbuilt and SMPS type, it shall also work on 230V AC supply.		
11	The system should support have rack, wall or table top mounting options.		
12	The system should have an ISDN Digital platform and shall be compatible with ISDN PRI line of Local Service Provider.		
13	Multiple systems at different locations should be able to connect with each other without any link licenses.		
14	The system should have built-in public address port and external music port.		
15	The system should be license-free to use third party SIP phones.		
16	The system should have built-in 15 participants conference i.e. 3 conferences of five parties each or 1 conference of 15 parties should be offered built-in with the platform.		

17	The system should have built-in auto attendant facility and should be able to answer minimum 5 calls simultaneously and should support dial-by-name.		
18	The system should have in-skin GSM card so that the multiple SIMs can be inserted on the GSM card plugged on to the PBX platform. Hence, the calls on GSM mobile can be routed through these SIMs and contribute in reduction of overall telecom bill. External device for GSM connectivity should not be mandatory.		
19	The system platform should always be ready for ISDN and VoIP. Only ISDN & VoIP cards must be added for functionality.		
20	The system should support SMPP protocol to send/receive SMS using in-skin GSM SIMs within system. Any software required to send/receive SMS should be quoted separately.		
21	The system should have at least 1 RS232 port for SMDR/PMS/CAS Interface.		
22	The system should have the functionality to be programmed through Analog telephone, Digital key phone and Ethernet without any external devices.		
23	The system should have a built-in remote maintenance facility. It should have the facility be programmed remotely over the internet without any modem required on the PBX side.		
24	The call ringing sequence should be programmable and have options such as simultaneous ring, hunting off, round robin and delayed simultaneous.		
25	Caller Line Identification (CLI) on Analog and digital/PRI trunks should be built-in for both DTMF and FSK telephone instrument.		
26	Detail reports of all system parameters should be generated through the SMDR port of System. External third party Billing software should not be required for basic report generation. Reports should be directly saved in PDF format.		
27	The system should be QSIG ready (for PRI) for networking and feature transparency between two or more exchanges.		
28	Each port of the system should be programmable. It should have programmable features port-wise/extension-wise.		
29	The system should 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.		
30	The system should have built-in web based software programming tool for system administration.		
31	Access codes, system timers and access to features should be programmable.		
32	Storage of outgoing, incoming and internal call reports should be generated on SMDR port of the system. It should also be available online through Ethernet Port.		
33	The system should have built-in outgoing Call Log buffer of 6000 calls, incoming call log buffer of 5000 and call log buffer of 1000 internal calls.		
34	Voice guided auto-attendant should be built-in.		
35	System should support dial form the directory. There should be minimum 900numbers possible and shall also possible to dial it as an abbreviated number.		

36	Features given to an extension should be accessed from any other extension by dialing the feature access codes.		
37	The system features should have class of service, night service, conference, auto diagnostic etc. Class of service shall be unrestricted, STD restricted and semi restricted.		
38	The System should support SMS to Email and Email to SMS Feature without use of any additional software.		
39	The system should support SMS on NO reply.		
40	The system should have built-in SNTP client.		
41	The system should have features as CLI based routing, call duration control, least cost routing i.e. time, number or combination of both.		
42	Extension features should 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.		
43	The system should have a conversational recording in the mail box should be available with voice mail system card of System. Conversation recording should be possible on Analog/Digital/IP deskphones as well as Mobile SIP Smartphones (Android/iPhone).		
44	The system should have security dialer. i.e., system should have provision to any sensor such as Glass break sensor; fire sensor etc. should connect directly to the system. When sensors get activated system will dial out the pre-programmed number and deliver prerecorded emergency message as well as request for confirmation.		
45	The system should support following features of IP telephony: Dynamic DNS, Registrar Server, Proxy Server, Presence Server, NAT and STUN, VoIP codec G.711u, G.711a, G.723, G.729		
46	The system should provide IP functionality to support IP extensions and trunks over SIP protocol. It should be possible to support IP trunks and extension with the single VoIP expansion card. Further expansion of VoIP channels should be possible with an expansion card.		
47	The system should support 50 IP Users and 16 VoIP (SIP) Trunks.		
48	Varied type of open SIP/IP Terminals such as IP Phone, SIP softphone and Mobile SIP Client should be supported.		
49	The VoIP card should have 16/8 channels per card and such multiple cards shall be used to increase the number of VoIP channels.		
50	The manufacturer should also have an application for Android and iPhone so that the enterprise mobility can be extended for the Smartphone users.		
51	No licensing for Android/iOS based smartphones as office extensions. Proprietary mobile softphone client should be available.		
52	Supplied IP Phones and PC based soft phones should be of the same manufacturer.		
53	IP functionality of the system should be in the form of in-skin interface card and can be inserted in the any slots on the platform.		

54	The system should integrate in-skin voice mail card with 72 hours of storage capacity and dedicated mailbox for each extension. It should support expandable storage capacity up to 576 hours.		
55	The system should have a conversational recording in the mail box available with voicemail card of the system. Conversation recording should be possible on Analog/Digital/IP as well as Mobile SIP Smartphones (Android/iPhone).		
56	IP phone operational functionality should be same as Digital Extension of PBX.		
57	The system should support SNMP.		
58	The system should support video conferencing over ISDN PRI.		
59	The system should have capability to support video call over IP.		
60	The system should be 19" rack-mountable.		
61	The system should be IPv6 complaint.		
62	System should be Fully operational between Temperature 0°C to +45°C		
63	<b>System should have in-skin Voicemail System with following features:</b>		
	Attend as much as 16 calls simultaneously with flexibility of routing callers to desired extension or delivering information depend upon the selection		
	Dial-by-Name to reach the intended user directly without knowing/remembering extension number		
	Selectively allocate voicemails to users with the flexibility of customizable mailbox size and greetings for All/Selective users		
	Group mailbox to share messages between departmental groups		
	Anywhere access to voice mail with just a phone call		
	Password protected secured voice mail access		
	Record important conversations for future reference and record maintenance		
	Redirection of voice mails to another extension in case of non-availability		
	Tag voice mails while Forwarding Messages to Another Mailbox		
	Broadcast voice message to a group of personnel, at a go		
	Distribution lists for delivery of voice mails to different set of users or groups		
	Message wait indication via ring, change in dial-tone, voice message or message wait lamp		
	Notification of a new voice mail via email alert or a phone call		
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 should comply following feature list:		
•	Abbreviated Dialing (Global & Personal)	
•	Access Codes (Programmable)	
•	Account Codes (Forced)	
•	Alarm-Multiple	
•	Alarms	
•	Alarm-Snooze	
•	Allowed and Denied Lists	
•	Alternate Number Dialing	
•	Analog Input Port (AIP)	
•	Analog Output Port (AOP)	
•	Anonymous Call Rejection (SIP)	
•	Auto Call Back (Busy, No Reply)	
•	Auto Redial	
•	Auto-Attendant	
•	Automated Control Applications	
•	Background Music (DKP and SL T)	
•	Backup SMDR	
•	Backup-System Configuration	
•	Backup-System Software	
•	Barge-in	
•	Boss Ring	
•	Call Budget on Extensions	
•	Call Budget on Trunks	
•	Call Chaining	
•	Call Cost Calculation	
•	Call Duration Control	
•	Call Follow Me	
•	Call Forward (Busy, No Reply, Dual Ring and to External Number)	
•	Call Park (General and Personal Orbit)	
•	Call Pick Up (Group and Selective)	
•	Call Progress Tones (Programmable)	
•	Call Splitting	
•	Call Taping	
•	Call Transfer (Screened, On Busy, While Ringing, Trunk to Trunk)	
•	Calling Line Identification & Presentation (CLIP)	
•	Calling Line Identity Restriction (CLIR)	
•	Cancel All Station Features	
•	Class of Service (COS)	
•	CLI based Routing	
•	Closed User Group (With/Without Exchange ID)	
•	Computer Telephony Integration (CTI)	

• Conference Dial-in		
• Conference – Multiple Participants		
• Conflict Dialing		
• Continued Dialing		
• Conversation Recording		
• Date and Time Format		
• Daylight Saving Time (DST)		
• Direct Inward Dialing (DID)		
• Day-Night Mode		
• Department Call		
• Digest Authentication (on SIP)		
• Direct Dialing-In (DDI on T1/E1/PRI)		
• Direct Inward System Access (DISA)		
• Direct Outward System Access (DOSA)		
• Direct Station Selection Console		
• (DSS Console)		
• Distinctive Rings		
• Do-Not-Disturb (DND)		
• Do-Not-Disturb (Remote)		
• Door Phone Connectivity (ETERNITY PE)		
• Dual Ring		
• Dynamic DNS (DDNS)		
• Dynamic Lock (Manual)		
• E-mail Notification (VMS)		
• Embedded Registrar and Proxy Servers (SIP Server)		
• Emergency Calls Detection and Reporting		
• Emergency Conference		
• Emergency Number Dialing		
• External Call		
• External Call Forward (ECF)		
• External Music Port (AIP)		
• Fax over IP (T .38 Relay and Pass-Through)		
• File Transfer Protocol		
• Flexible Numbers (Up to 6 Digits)		
• Forced Answer		
• Help Desk		
• Hold		
• Hot Desking		
• Hot Outward Dialing (With/Without Number & Delay)		
• Hotline (Immediate and with Delay)		
• Hunting/User Group		
• Incoming Call Management		
• Installation Wizard		
• Internal Call		
• Internal Call Restriction		

• Interrupt Request		
• Last Caller Recall		
• Last Number Redial		
• Least Cost Routing (Number, Time and Service		
• Provider to Service Provider, Carrier Pre-Selection)		
• Live Call Screening (VMS)		
• Live Call Supervision		
• Logical Partitioning		
• Maturity (Polarity Reversal, Delay, CPD)		
• Meet Me Paging		
• Menu based Command (DKP)		
• Message Wait Indication		
• Missed Calls		
• Music-On-Hold		
• Mobile Port (GSM/3G Port)		
• Multi-Stage Dialing		
• Mute		
• Name Programming (Station, Trunk)		
• NAT and STUN (VoIP)		
• Network Selection (GSM)		
• Off-Hook Alert (DKP)		
• Online SMDR		
• Operator (Single, Multiple)		
• Override		
• Paging (Internal and External)		
• Peer-to-Peer Calling		
• Priority (Intercom and Trunk)		
• Privacy		
• Programming the System		
• (Using SL T , DKP , Ethernet Port)		
• Public Address System Port		
• Quick Dial		
• Raid		
• Real Time Clock		
• Region Selection		
• Remote Alarm		
• Remote Call Forward		
• Remote Programming		
• Routing Group		
• Return Call to Original Caller (RCOC)		
• Room Monitor		
• RS232C Port		
• SMDR Posting (Call Accounting System Interface)		
• SMS Gateway		
• SMS Server		



	• Security Dialing and Reporting		
	• Self-Ring Test		
	• Selective Trunk Access		
	• SIP and RTP QoS (VoIP)		
	• Station Groups		
	• Station Message Detail Record - 12000 Records)		
	• Station Name		
	• System Activity Log and Display		
	• System Administrator (SA) Mode		
	• System Engineer (SE) Mode		
	• System Fault Log		
	• System Security (Password)		
	• Time Tables		
	• Time Zone Display		
	• Toll Control		
	• Trunk Access Group		
	• Trunk Auto Answer		
	• Trunk Connectivity (BRI)		
	• Trunk Connectivity (GSM/3G)		
	• Trunk Connectivity (T1/E1/ISDN PRI)		
	• Trunk Connectivity (TWT or CO)		
	• Trunk Connectivity (VoIP)		
	• Trunk Reservation		
	• Upgrading the Software		
	• User Absent/Present		
	• Virtual Stations		
	• Voice Message Applications		
	• Voice Prompts for T ones		
	• Walk-In Class of Service (Single/Multiple calls)		
	• Web based Programming		
	• GSM Trunk Connectivity		
	• Automatic Call to Missed (Predefined) Calls on Trunks		
	• Routing of calls to only permissible legal networks (Logical Partitioning)		
	• SMDR though Ethernet Port		
65	<b>Mobile Soft client features for android/iPhone :</b>		
	• Shall be installed on android OS 2.2 or later		
	• Shall be installed on IOS 7 or later		
	• Comprehensive Call Management		
	• One-Touch Access to PBX Features		
	• Corporate Directory Integration		
	• Video Calling		
	• Favorites		
	• Presence Sharing and Instant Messaging		
	• Voice Mail Access		

	<ul style="list-style-type: none"> <li>• Conversation Recording</li> </ul>		
	<ul style="list-style-type: none"> <li>• Call management – call hold, transfer, forward, DND and intercom</li> </ul>		
	<ul style="list-style-type: none"> <li>• Multiparty audio conferencing</li> </ul>		
	<ul style="list-style-type: none"> <li>• Blind transfer</li> </ul>		
	<ul style="list-style-type: none"> <li>• Menu options - Call Pickup, Paging, Message wait, Call Retrieve, Alarm and Reminder, Dynamic Lock, Dial-In Conference, CLIR, Room monitoring &amp; Call supervision</li> </ul>		
	<ul style="list-style-type: none"> <li>• Dial by extension</li> </ul>		
	<ul style="list-style-type: none"> <li>• Call Screening - ACB, Forced Answer, Global Hold, General Call Park, Call Chaining,IR &amp; Barge-IN</li> </ul>		
	<ul style="list-style-type: none"> <li>• Multiple call support</li> </ul>		
	<ul style="list-style-type: none"> <li>• SMS over IP</li> </ul>		
	<ul style="list-style-type: none"> <li>• One touch transfer</li> </ul>		
	<ul style="list-style-type: none"> <li>• Wi-Fi to cellular handover</li> </ul>		
	<ul style="list-style-type: none"> <li>• Multiple language support</li> </ul>		
	<ul style="list-style-type: none"> <li>• Call toggle</li> </ul>		
	<ul style="list-style-type: none"> <li>• Auto call back</li> </ul>		
	<ul style="list-style-type: none"> <li>• Auto redial</li> </ul>		
	<ul style="list-style-type: none"> <li>• Forced answer</li> </ul>		
	<ul style="list-style-type: none"> <li>• Open a door</li> </ul>		
	<ul style="list-style-type: none"> <li>• Call logs</li> </ul>		
	<ul style="list-style-type: none"> <li>• Missed calls notification</li> </ul>		
	<ul style="list-style-type: none"> <li>• DND override</li> </ul>		
	<b>UC client Features:</b>		
	<ul style="list-style-type: none"> <li>• Windows Desktop based UC Client for Seamless Collaboration</li> </ul>		
	<ul style="list-style-type: none"> <li>• Corporate Directory Integration</li> </ul>		
	<ul style="list-style-type: none"> <li>• 1000 DSS &amp; 600 BLF keys for Monitoring</li> </ul>		
	<ul style="list-style-type: none"> <li>• Popup Window for Incoming Message and Call</li> </ul>		
	<ul style="list-style-type: none"> <li>• Presence Sharing and Instant Messaging</li> </ul>		
	<ul style="list-style-type: none"> <li>• Video Calling</li> </ul>		
	<ul style="list-style-type: none"> <li>• Drag and Drop Conference</li> </ul>		
	<ul style="list-style-type: none"> <li>• Contact Grouping</li> </ul>		
	<ul style="list-style-type: none"> <li>• Video call</li> </ul>		
66	<ul style="list-style-type: none"> <li>• Voicemail</li> </ul>		
	<ul style="list-style-type: none"> <li>• Favourites</li> </ul>		
	<ul style="list-style-type: none"> <li>• Call management – call hold, transfer, forward, DND and intercom</li> </ul>		
	<ul style="list-style-type: none"> <li>• Multiparty audio conferencing</li> </ul>		
	<ul style="list-style-type: none"> <li>• Blind transfer</li> </ul>		
	<ul style="list-style-type: none"> <li>• Menu options - Call Pickup, Paging, Message wait, Call Retrieve, Alarm and Reminder, Dynamic Lock, Dial-In Conference, CLIR, Room monitoring &amp; Call supervision</li> </ul>		
	<ul style="list-style-type: none"> <li>• Handover to external number</li> </ul>		
	<ul style="list-style-type: none"> <li>• Smart directory access</li> </ul>		
	<ul style="list-style-type: none"> <li>• Dial by extension</li> </ul>		

• Call Screening - ACB, Forced Answer, Global Hold, General Call Park, Call Chaining, Call Recording, IR & Barge-IN		
• Multiple call support		
• SMS over IP		
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