First Security Islami Bank Limited
ICT Division
Head Office
Plot#12, 2nd & 3rd Floor, Main Road.
Block # A, P.S- Badda,
Bashundhara R/A, Dhaka-1229.
Web: www.fsiblbd.com; email:
ictp@fsiblbd.com

Ref: FSIBL/HO/ICT/**595**/2019 Date: 31.10.2019

Notice: Inviting Tender for supplying, Installing & commissioning of "IP Telephony and Video Conferencing System" for First Security Islami Bank Limited.

Sealed Tenders are hereby invited by First Security Islami Bank Limited from reputed Manufacturers / Local Agents / Companies / Dealers/ Partners/ Distributors/ Resellers for supplying, Installing & commissioning of IP Telephony and Video Conferencing System for First Security Islami Bank Bangladesh Limited under the following terms and conditions:

1. Technical Specification of IP Telephony System:

General

- The system should be based on server-gateway architecture running on Linux OS supporting IP, Analog and/or Digital Extensions.
- System shall be non-blocking and open standard (based on H.323 and SIP). It should support Active-Active Redundant Servers, in load sharing architecture, able to switchover to other server in case of failure of one server.
- **The servers should be of industry standard makes like DELL/HP/IBM/CISCO etc with redundant power supply. The proposed server should be capable of supporting at least 4000 from day one and should be expandable in future. The expansion should not require any additional hardware.**
- The solution should support at least 128 DSP channels on each gateway.
- The system should be capable of deployment on virtualized platforms like VMWare/Xen etc.
- System should provide call control, mobility, IM and presence, and Messaging, centralized licensing in a single server.
- System should support secondary/redundant server for additional capacity and resiliency.
- System should have fax on demand; if required it may be provisioned through third party Fax application.
- The system should support standards-based multi-site networking, using QSIG, H.323 trunks or advanced networking, to interoperate with other PABX's, allowing feature transparency.
- System should be able to provide backup/redundancy options in case of failure of one server.
- The system should support at least 100 remote sites on the same platform through gateways.
- System should be able to provide centralized voicemail with the option of Distributed centralized voicemail in case of connectivity failure.
- The system should support BRI/ PRI/ T1/ E1/Analog/IP/SIP Trunks.
- The server should have in-built atleast 2 nos auto-sensing 10/100/1000 Mbps LAN.
- The gateway system should support X.21/V.35 WAN Interface.
- The system should support internal MOH (Music on Hold), which should be uploaded using the .Wav file and should have an audio input port for external MOH connectivity.
- The system should be 19" rack mountable.
- The equipment quoted by bidder must be SIP compliant.
- **The OEM must feature in the Leaders segment of the Gartner Magic Quadrant for Corporate Telephony published in any of last three years.**
- **For seamless integration, IP Telephony and Video Conferencing Solution should be same from OEM**
- **Proposed system should have minimum 4000 user capacity from day one without changing hardware.**
- **Proposed solution GB should support non business impacted failover and there should not be any capacity or features degradation after failover. Also none of the phone should get rebooted.**
- **Proposed solution should support gateway with duplicated power supply and also should support IPV6.**
- **Proposed solution should provide video conferencing system with 80 participants in HD quality.**

1.1 Smartphone & Tablet Capabilities

The proposed hybrid-IP telephone system should have the ability to be used/accessed from a smartphone and/or a tablet device.

- The Softphone should provide full call control from an iPhone or Android powered smartphone.
- Make and receive phone calls and instant messages, host and attend audio conferences.
- See employee availability via presence, and use Geo-tracking to determine the location in the field.
- All of this is done using the corporate directory, so there are no personal cell phone numbers will be involved.
- The Softphone application should be downloadable from Google Playstore or Apple iTunes without any additional cost for any number of device.

1.2 Security Features:

- **System should support AES 256 encryption to ensures that all data sent by either the system or Manager cannot be 'read' by anyone else, even another copy of Manager**
- **Systems should support Secure Real-Time Transport Protocol (SRTP) for the application of additional encryption and or authentication to VoIP calls (SIP and H.323).**
- **Proposed SIP trunk should be secure and also support Media transcoding for better operability. Also it should support Quality-of-service (QoS) and bandwidth management.**

1.3 Telephony Features

Besides the normal Telephony features, the system should support the features like Absent Text, Call Coverage, Call Forwarding, Call Hold, Call Intrude, Call Park, Call Pickup, Ring Back When Free, Suspend Call Waiting, Reclaim Call, Toggle Calls, Account Codes, Call Barring, Authorization codes, Bridged Appearance, Group Paging, Group Listen, Hot Desking, Mobile Twinning, Least Cost Routes, Alternate Route Selection, Flexible numbering Schemes, Time of Day and Date Routing of Calls, Call Recording, Maximum Call Length, PIN Restricted Calling, Time Profiles, Queuing, Queue announcements, Call Detail Recording , SMDR

1.4 Data Communication Features:

- System should have in built-in DHCP Server, which should be able to give IP Addresses to the endpoints.
- System should support built-in Remote access server (RAS) functionality.
- System should support Diffserv for QoS (Quality of service) for the voice packets traveling over data networks
- System should support NAT
- System should support LDAP (Local Directory Access Protocol)

1.5 Terminal Support :

System should support the following type of terminals

- Analog or Digital Phones
- IP Hardphones
- IP Softphones
- Wireless IP Phones (802.a/b/g/n)
- 3rd party SIP telephones

1.6 Extensions and Trunks:

- System should be able to support up-to 4000 extensions in any combination.
- System should have built-in H.323 gatekeeper functionality without the need to put any additional hardware.
- System should support SIP trunking to Internet Telephony Service Providers, allowing non-SIP phones to make SIP calls.
- System should support following types of trunks: Analog Trunks, PRI, BRI, T1/ E1/ E1R2

1.7 Voicemail Features

- System should have inbuilt voice mail system.
- Voicemail to email option should be available
- System should support unified messaging with Microsoft Exchange or any IMAP compliant email application.

 Signature & Seal of the bidder

- System should support voicemail access through web-browser
- External Fax server integration should be possible.
- VM should support text-to-Speech functionality
- VM should support Dial-by-Name functionality
- VM should support Auto Attendant
- IVR functionality should be available

1.8 Conferencing Features:

- The system should have built-in 128 party Meet-Me conferencing bank from day 1 and should be expandable to 256 party.
- Multiple conferences with variable number of users should be possible within each of the conferencing banks.
- System should be able to generate detailed reports about the conference.
- System should be able to send emails to all the participants giving them the conferencing details
- System should support PIN based security for conference calls.

1.9 Video Capability:

- System should have IP Soft-phone capability with video support
- System should be able to provide board-room units as extensions to the system support various types of conference rooms.

1.10 Call Recording:

- System should have capability to automatically as well as manually record and store calls into any voicemail box or a central database, for later retrieval, sorting, searching through a web-based browser interface from day one.
- Recordings should be able to be made on the basis of: User ID, Account code, Hunt Group, Caller ID, Incoming call route, Time profiles
- System should support automatic deletion of oldest recordings, if needed.
- System should support G.726 16kbps ADPCM standard for compressing and storing recordings, providing the best compromise between CPU loading and storage space.

1.11 Mobility Support:

- System should support Mobile Twinning, enabling an extension and an internal/ external number to operate together as a single telephone. It should be possible to set external mobile devices as twinning targets, even if the primary extension is logged out/ unplugged.
- System should be able to provide 'work from home' features like telecommuting and VPN hard phones.

1.12 Presence/IM/thin-client Application

- System should support simplified call control features like tap to call, tap to conference etc.
- Visual voice mail, IM, presence
- Central directory access
- Support for Android and iOS devices (smartphones, tablets)

1.13 Wireless Support:

- System should support wireless IP Phones which will work through the Access Points which are being used for Wireless Data network supporting 802.11a/b/g protocol.

1.14 Management utilities:

- System should be able to be configured and administered using a GUI based application
- System should support SNMP based network management
- In case SNMP management is not available, system should be capable of sending event notifications to up-to 3 email addresses, each with a different set of alarms

1.15 Desktop Phone Specifications (Type-1):

Capacitive minimum **7-inch full touch screen with Resolution**: 720x1280 , The desk phone should enable users to Audio mute, Video mute, headset, speaker, Message Waiting Indicator The full duplex speakerphone must allow for hands-free conversations, and it must have Power over Ethernet port which offers convenient access with any network.

The Phone must have call logs and address books, and visual message waiting indicator. The phone should have built-in dual Ethernet 10/100/1000 port which allows PCs to connect to the network.

Detailed Feature Description

Audio

• Supported codecs: G.722, G.711, G.729, G.726, H.264
• Full-duplex speakerphone

Call Handling

• SIP/H.323
* Mute/Unmute
* Last number redial
* Transfer
* Forward

User Interface

• Operating System: **Android/OEM Specific Connection**
• Dual 10/100/1000 Ethernet ports
• Wireless access point mode, Hotspot, Wi-Fi 802.11a/b/g/n/ac, Bluetooth 4.2

1.16 Desktop Phone Specifications (Type-2):

Capacitive minimum 5-inch screen with Resolution: 720x1280 , The desk phone should enable users to Audio mute, Video mute, headset, speaker, Message Waiting Indicator The full duplex speakerphone must allow for hands-free conversations, and it must have Power over Ethernet port which offers convenient access with any network.

The Phone must have call logs and address books, and visual message waiting indicator. The phone should have built-in dual Ethernet 10/100/1000 port which allows PCs to connect to the network.

Detailed Feature Description

Audio

• Supported codecs: G.722, G.711, G.729, G.726, H.264
• Full-duplex speakerphone

Call Handling

• SIP/H.323
* Mute/Unmute
* Last number redial
* Transfer
* Forward

User Interface

• Operating System: **Android/OEM Specific Connection**
• Keypad with the standard keys 0-9, *, #

Connections

• Dual 10/100/1000 Ethernet ports
* Headset jack (RJ9 connector)

1.17 Desktop Phone Specifications (Type-3):

Minimum 2.8” (diagonal) Grascale/color display with Physical Buttons & Status Indicators, The full duplex speakerphone must allow for hands-free conversations, and it must have Power over Ethernet port which offers convenient access with any network. The Phone must have call logs and address books, and visual message waiting indicator.

Detailed Feature Description

Audio

• Supported codecs: G.722, G.711, G.729, G.726, H.264
• Full-duplex speakerphone

Call Handling

• SIP/H.323
* Mute/Unmute
* Last number redial
* Transfer
* Forward

User Interface

• Multiple line phone with four red/ green line/feature indicators around display
• Keypad with the standard keys 0-9, *, #

Connections

- **Dual 10/100 Ethernet ports (PoE Class IEEE standard)**

1.18 Desktop Phone Specifications (Type-4):

The desk phone should enable users to transfer, mute, forward and place calls on hold as well as initiate ad-hoc conferences by a dedicated conference feature button. The full duplex speakerphone must allow for hands-free conversations, and it must have Power over Ethernet port which offers convenient access with any network.

The Phone must have call logs and address books, and visual message waiting indicator. The phone should have built-in dual Ethernet 10/100 port which allows PCs to connect to the network.

Detailed Feature Description

Audio

• G.722 wideband audio
• Full-duplex speakerphone

Call Handling

• SIP/H.323
• Single line, 2 calls/line operated with “flash” key
* Mute/Unmute
* Last number redial
* Transfer
* Forward

User Interface

• Monochrome/color display with minimum two rows
• Context-sensitive soft keys
• Status indicators

Connections

- **Dual 10/100 Ethernet ports (PoE Class IEEE standard)**
- Headset jack (RJ9 connector)

Signature & Seal of the bidder

1.18 Operator/Master IP Phone Specifications (Type-5)

Minimum 3.5” (diagonal) Grayscale/Color display with Physical Buttons & Status Indicators, The full duplex speakerphone must allow for hands-free conversations, and it must have Power over Ethernet port which offers convenient access with any network.

The Phone must have call logs and address books, and visual message waiting indicator. The phone should have built-in dual Ethernet 10/100/1000 port which allows PCs to connect to the network.

Detailed Feature Description

Audio

• Supported codecs: G.722, G.711, G.729, G.726, H.264

• Full-duplex speakerphone

Call Handling

• SIP/H.323

* Mute/Unmute

* Last number redial

* Transfer

* Forward

User Interface

• Multiple line phone with four red/ green line/feature indicators around display
• Keypad with the standard keys 0-9, *, #

• Minimum 24 programable button expansion unit
• Soft operator console

Connections

• Dual 10/100/1000 Ethernet ports (PoE Class IEEE standard)

Signature & Seal of the bidder

1.19 Network Switch

Item Name	Required Specification	Unit	Qty	Bidder's Response
Network Switch: 24 x 1G ethernet PoE	• Brand: CISCO/Juniper • 19” rack mountable - Stackable Switch / Chassis • Switching Mode supported: Store and forward • Simultaneously deliver up to 15.4 watts of standards-based 802.3af Class 3 PoE to a maximum of 24 ports with power budget of 370 watts. • 2 X Small Form Factor Pluggable 10/100/1000 Copper Transceiver ModulePacket Switching Capacity - 128 Gbps or more • L2 Throughput - 95 Mpps or more • OS should support individual process (eg ssh , snmp, telnet, dhcp etc) restart to prevent reboot in case of Software Process Crash by running processes on top of Kernel.	Nos	25	

1.20 Offline UPS for VC

Item Name	Required Specification	Unit	Qty	Bidder's Response
Offline UPS	Uninterruptible Power Supply (UPS)	Nos	9	
Brand	Mention Please			
Model	To be Submitted by the bidder			
Country of origin	Mention Please			
Country of Manufacture	Mention Please			
Country of shipment	Mention Please			
Capacity	1200 VA			
Backup	At least 30 minutes battery backup at full (100%) load			
Battery Brand	Mention Please			

Signature & Seal of the bidder

1.21 Video Conferencing System for Head Office:

Item Name	Required Specification	Bidder Remarks
Brand	To be mentioned by the bidder	
Model	To be mentioned by the bidder	
Country of Origin	EU/USA	
Country of Manufacturer	To be mentioned by the bidder	
Warranty	2 year warranty	
Part Number	Part Number	
ISO/FC/CC or Any Other Certificate	To be mentioned by the bidder	
Communications	IP: H.323, SIP (RFC 3261), Bit rate: H.323, SIP: up to 6/12 Mbps, Presence Server Support	
Video	2 simultaneous 1080p 60fps channels: camera + content, H.263, H.263+, H.263++, H.264, H.264 High Profile, H.264 SVC, Dual: H.239 (H.323); BFCP (SIP); Multi-Stream Switching(MSS) video	
Camera	**Video Format: 1080p60/50/30/25** **Field of View: H/V 74° / 44°** **Automatic speaker tracking up to 20feet** **Image Device: CMOS 1/2.8''**	
HDMI Input Formats	1920 x 1080p @ 25, 30, 50, 60fps1280 x 720p @ 25, 30, 50, 60fps 720 x 576p @ 50fps 720 x 480p @ 60fps 640 x 480p @ 60fps	
PC content encoding formats:	Up to 1080p @ 60fps	
HDMI output formats	1920 x 1080 @ 25, 30, 50, 60fps1280 x 720 @ 50, 60fps	
Content Sharing / Web Collaboration	Dual video: H.239 (H.323); BFCP (SIP)	
Auxiliary Camera Interface	VISCA RS232 for PTZ control (Mini-DIN 8-pin)	
Recording	• Should supports Recording (with dual display systems, both displays show the same images while recording),**From Day One** • Recording formats: MP4 file; Time watermark on recorded files; Digital Signature for recorded files	
Audio	• Acoustic echo cancellation• Adaptive post filtering• Audio error concealment• Automatic Gain Control (AGC)• Automatic noise suppression	
Microphone	• 360° range• Frequency response: 50 – 22,000 Hz • Stereo audio with 3-way POD work in a cascading mood. • Sensitivity: **-37dB +/-3dB** • Signal to Noise Ratio: **>68dB** • 2 Unit Microphone Pod **from day one.**	
Video inputs	1 x HDMI 1	
Video outputs	2 x HDMI for dual display	
Audio inputs	1 x HDMI1 x Microphone Array Pod (RJ-11)1 x 3.5 mm (1/8") line in jack for audio source (analog / digital with mini-TOSLINK adapter); 3 x Additional HDMI via switcher	
Audio outputs	2 x HDMI1 x 3.5 mm (1/8") line out jack (analog / digital with mini-TOSLINK adapter)	
USB Ports	1 x USB2, 1 x USB3, Supports recording, firmware upgrade, USB to serial adapter for control, limited headsets and USB2 cameras	
Network Features	IPv4 and IPv6 simultaneous support• Auto Gatekeeper discovery• QoS support: IP Precedence, IP Type of Service (ToS), Differentiated Services (diffserv)• Bandwidth adaptation for high quality over unmanaged networks• DTMF tones: H.245, RFC2833• SNTP date and time synchronization	
IP adaptive packet management	Flow control Packet Loss based down-speeding Packet Loss recovery policies	
Touch Control Device/Application	A control application/device with a multi-touch interface like the Apple iPad/iPhone is desired **from day one**. This interface should enable the user to: • Dial an address with a list of the recent outgoing, incoming or missed calls or Access the company directory and place a call from the directory • Control the VC Room Camera (PTZ), mute microphone, change volume, set DND, start and stop presenting. • Inviting another participant by either dialing by address (IP, E.164 or SIP URI) or by accessing the company directory	
Network Interfaces	2 x 10/100/1000 Base-T full-duplex (RJ-45)	
Firewall Traversal	Auto NAT discovery HTTP and STUN• H.460.18, H.460.19• "Keep Alive" packet time configurable	
Camera Interface	VISCA RS232 for PTZ control (Mini-DIN 8-pin)	
Security Features	Embedded encryption • H.323 security per H.235v3/Annex D• SIP security per SRTP and TLS 1.2, enciphering up to AES-256• SIP "Digest" authentication (MD5)• RTP media enciphering with AES-256• Public-Key including Diffie-Hellman (2048 bits), RSA (up to 4096 bits)• Cryptographic hash (up to SHA-512) • Web access via HTTPS• API access via SSH• Selective enable / disable of IP features• White list support; Enhanced Access Security Gateway support	
Web Management	• Configuration, call, diagnostics and management functions are accessible with Internet Explorer 8.0+, Firefox 3.6+, Safari 5.0+, Chrome 11.0+, Opera 11+	

1.22 Video Conferencing System for Zonal Office & Training Institute:

Item Name	Required Specification	Bidder Remarks
Brand	To be mentioned by the bidder	
Model	To be mentioned by the bidder	
Country of Origin	EU/USA	
Country of Manufacturer	To be mentioned by the bidder	
Warranty	2 years warranty	
Part Number	Part Number	
ISO/FC/CC or Any Other Certificate	To be mentioned by the bidder	
Communications	IP: H.323, SIP (RFC 3261) Bit rate: H.323 SIP: up to 4/121 Mbps MCU compatibility: H.243, H.231 XMPP Presence Server	
Video	2 simultaneous 1080p 60fps channels: camera + content, H.263, H.263+, H.263++, H.264, H.264 High Profile, H.264 SVC, Dual: H.239 (H.323); BFCP (SIP)	
HDMI Input Formats	1920 x 1080p @ 25, 30, 50, 60fps1280 x 720p @ 25, 30, 50, 60fps720 x 576p @ 50fps720 x 480p @ 60fps640 x 480p @ 60fps	
PC content encoding formats:	Up to 1080p @ 60fps	
HDMI output formats	1920 x 1080 @ 25, 30, 50, 60fps1280 x 720 @ 50, 60fps	
Camera	**Video Format: 1080p60/50/30/25** **Field of View: H/V 74° / 44°** **Automatic speaker tracking up to 20feet** **Image Device: CMOS 1/2.8"**	
Recording	• Should Supports recording **from day one** (with dual display systems, both displays show the same images while recording),	
Audio	• Acoustic echo cancellation• Adaptive post filtering• Audio error concealment• Automatic Gain Control (AGC)• Automatic noise suppression	
Microphone	• 360° range• Frequency response: 50 – 22,000 Hz • Stereo audio with 3-way POD • Sensitivity: **-37dB +/-3dB** • Signal to Noise Ratio: **>68dB**	
Video inputs	1 x HDMI1	
Video outputs	1 x HDMI for main display1 x HDMI for second display	
Audio inputs	1 x HDMI1 x Microphone Array Pod (RJ-11)1 x 3.5 mm (1/8") line in jack for audio source (analog / digital with mini-TOSLINK adapter)	
Audio outputs	2 x HDMI1 x 3.5 mm (1/8") line out jack (analog / digital with mini-TOSLINK adapter)	
USB Ports	1 x USB2, 1 x USB3	
Network Features	IPv4 and IPv6 simultaneous support• Auto Gatekeeper discovery• QoS support: IP Precedence, IP Type of Service (ToS), Differentiated Services (diffserv)• Bandwidth adaptation for high quality over unmanaged networks• DTMF tones: H.245, RFC2833• SNTP date and time synchronization	
IP adaptive packet management	Flow control Packet Loss based down-speeding Packet Loss recovery policies	
Network Interfaces	2 x 10/100/1000 Base-T full-duplex (RJ-45) (2nd GLAN is optional)	
Firewall Traversal	Auto NAT discovery HTTP and STUN• H.460.18, H.460.19• "Keep Alive" packet time configurable	
Camera Interface	VISCA RS232 for PTZ control (Mini-DIN 8-pin)	
Security Features	Security Features• Embedded encryption3• H.323 security per H.235v3/Annex D• SIP security per SRTP and TLS 1.2, enciphering up to AES-256• SIP "Digest" authentication (MD5)• RTP media enciphering with AES-256• Public-Key including Diffie-Hellman (2048 bits), RSA (up to 4096 bits)• Cryptographic hash (up to SHA-512) • Web access via HTTPS• API access via SSH• Selective enable / disable of IP features• White list support	

 Signature & Seal of the bidder

1.23 Monitor/TV for Video Conferencing

Sl	Item Name	Item Description	Unit	Qty	Bidder's Response
1	**Display Panels /TV**			Nos	
	Brand:	Sony /Samsung /LG			
	Model:	Please Mention			
	Country of Origin:	Japan/ South Korea			
	Country of Manufacturer	Please Mention			
	Product Type	LED			
	Display	Screen Size & type: 55-inch		1	
		Screen Size & type: 43-inch		8	
		Resolution: 1080p			
		Aspect Ratio: 16:9			
		Viewing Angle: Min 178(89/89) Degree			
		PQI (Picture Quality Index): ≥1500			
		Motion Rate: ≥100			
		Color: Pur Color/ Dynamic Crystal Color			
	Audio	Speaker Type: Please Mention			
		Sound Output (RMS): ≥ 20W			
		Dolby Digital Plus: Please Mention			
	Connectivity	HDMI port: ≥ 2			
		USB port: ≥ 2			
		Ethernet (LAN) (RJ45)			
		Ex-Link (RS-232C)			
	Installation & Commissioning	Offer should include installation and commissioning with all required hardware, cable, splitter, stand/wall/ceiling mount kit andother accessories.			
	Warranty:	The vendor shall provide at least 3 years warranty			

Signature & Seal of the bidder

2.0 BOQ:

S/N.	Description of Items/Work	Qty.	Unit	Rate in Taka	Total Amount in Taka
1	**<u>IP PABX Communication Manager Server</u>** * **4000 User Capacity from day one** * Necessary IP Phone Licenses * Necessary voice mail, IVR licenses * 10 Channel Voice Call Recording License * 256 Party meet me conference bridge facility * 50 SIP trunk Channel license * Licenses for Recording function * Conference call recording * Vendor Need to Provide all necessary License, Server *** 10 FXO/CO port for BTCL connectivity**	1	Unit		
2	Top Management Touch Video IP Phone: (Type-1)	10	Unit		
3	Executive (Branch Manager) & Departmental Head Touch Video IP Phone Specifications:(Type-2)	216	Unit		
4	Branch Operation ManagerType-3)	190			
5	Basic IP Phone (Type-4)	610	Unit		
6	**Operator/Master IP Phone with Soft operator Console (Type-5)**	**1**	**Unit**		
7	IP Phone Local Power Adapter/PoE Injector (For Type 2 IP phones)	190	Unit		
8	IP Phone Local Power Adapter/PoE Injector (For Type 3 IP phones)	190	Unit		
9	Video Conference Endpoint for Head Office	1	Unit		
10	Video Conference Endpoint for Branch Office	8	Unit		
11	Video Gatekeeper to connect Mobile/Desktop user as a client	1	Unit		
12	Video Call Recording (For all location)	1	Unit		
13	55' Display Unit for Head office (Full HD, LED, Sony/Samsung)	1	Unit		
14	43' Display Unit for Branch office (Full HD, LED, Sony/Samsung)	8	Unit		
15	Network Switch: 24 x 1G Ethernet PoE	25	Unit		
16	Offline UPS (1200 VA)	9	Unit		
17	**<u>Installation & Commissioning Charges</u>** * Installation & Commissioning by Supplier * Installation & training at end user level (All over Bangladesh) * Two years OEM Support (After having Installation Completion Certification) * Installation Materials included	Lot	Job		

3.0 Qualification Criteria :

3.1 The OEM must feature in the Leaders segment of the Gartner Magic Quadrant for Corporate Telephony published in years 2018.

3.2 Warranty of IP Telephony should be 2 year warranty

3.3 ISO & FCC Certificate of OEM

3.4 Supplier should have qualification having minimum Fifty (50) lac valued Work Order (in one lot) of IP Telephony / Video Conferencing in any reputed organization in Bangladesh

3.5 Supplier should have sufficient storage capacity for future requirement (at least 02 years)

3.6 IP Telephony Solution & Video Conference Solution should be same OEM

3.7 Bidder’s advance collaboration certificate and collaboration certified engineer will be preferable

3.8 OEM Should have Registered office in Bangladesh

Signature & Seal of the bidder

4.0 Terms and Conditions:

4.1 The intending bidders have to apply in their letter head pad and must submit documentary evidence like VAT registration Certificate, Up to date Trade License & TIN Certificate, Authorized certificate for delivering of the item (if required), Certificate in support of their past experience and specialization in the field. On being satisfied with documents submitted by the applicant, Interested bidder may submit tender schedule along with a payment order of Tk.5,000/- (Five Thousand) only (non-refundable) from any scheduled bank in favor of First Security Islami Bank Limited. Tender schedule should be submitted in a tender box kept at the ICT Division of the bank (Address: ICT Division, (3rd Floor), Plot#12, Block#A, Main Road, Bashundhara R/A, Dhaka-1229) on or before 3:00PM 13-10-2019. The bidder have to copy or download this tender documents from the website: www.fsiblbd.com and place them on their own letterhead to submit their bid.

4.2 An original and one copy of the Offer duly marking **"Original Offer Technical"** and **"Copy of the Offer Technical"** in an envelope and **"Original Offer Financial" and "Copy of the Offer Financial"** in another envelope should be submitted separately at the time of tender submission with authentication by the Tenderers. **Combination of Technical and Financial Offer will be disqualified.** The bid form must be filled in through computer printer or in typing without overwriting and without any erasing and modifications and when completed shall contain all the required information.

4.3 5% (Five percent) of the total tender amount must be submitted along with the tender as performance security in favor of First Security Islami Bank Limited in the form of Bank Draft/ Bank guarantee from any scheduled Bank of Bangladesh preferably from First Security Islami Bank Limited without which the Tender shall be rejected outright.

4.4 The costs of complete installing necessary Software/application as required, testing, commissioning, delivering to directed site and admissible VAT, excise duty, subsidiary duty, import duty, ATV, AIT etc. all types of taxes and revenues of the government and other regulatory authorities along with time value of money up to settlement of bills taking Clearances from the end user of the bank.

4.5 The supplier will have to submit the bill after supplying, Installing and commissioning of all equipment with delivery challan. The bill will be paid after getting the satisfactory certificate from installation point & ICT Division, FSIBL, Head Office expressing clearly that the end user has no objection.

4.6 The Bank Authority reserves the right to -

1. Explain or clarify the terms of this tender notice in its own way,
2. Bring necessary changes in the notice
3. Increase or decrease the tender quantity
4. Reject the lowest
5. Reject any or all bids
6. Select any bidder deems fit and proper by them

4.7 The bank authority can perform all the above things without assigning any reason. The bidder/supplier shall have no right to challenge the decision of the Bank Authority in any court of law or to any arbitrator.

Signature & Seal of the bidder