

"औद्योगिक, पर्यटकीय र हरित नगर, स्वच्छ, सुरक्षित र समृद्ध नगर"

दमक नगरपालिका

नगर कार्यपालिकाको कार्यालय



www.damakmun.gov.np info@damakmun.gov.np

मिति: २०८०-१-२८



श्री सम्पुर्ण सूचीकृत फर्महरु

विषय: दर रेट पेश गर्ने सम्बन्धमा।

उपरोक्त सम्बन्धमा यस दमक नगरपालिकाका तथा वडा कार्यालयहरूमा IP Telephone जडान गर्नुपर्ने भइकाले यसै पत्र साथ उल्लेखित स्पेसिफिकेशन अनुसारको सामाग्रीहरूको दररेट मिति २०८०-०२-४ गते १२ बजे भित्र दमक नगरपालिकामा पेश गर्नुहुन अनुरोध छ।पेश हुन आएका दररेट सोहि दिन १ बजे खोलिने जानकारीको लागि समेत अनुरोध छ।

आवश्यक कागजातहरु:

- १. फर्म दर्ता प्रमाणपत्रको प्रतिलिपि
- २. मुअकर दर्ता प्रमाणपत्रको प्रतिलिपि
- ३. अघिल्लो आ.व.को कर चुक्ता
- ४. दररेट समावेस भएको कागजात
- ५. सामाग्रीहरुको Catalogue. (Brand, Model No. Specification आदी सम्पुर्ण उल्लेख भएको हुन पर्ने छ।
- ६. १ बर्षको Warranty Declaration.
- **9.** Manufacture Authorization

यज्ञराज निरौला बरिष्ठ प्रशासकीय अधिकृत

ख**राराज निरौला** वरिष्ठ प्रशासकीय अधिकृत

							IGABIT IP PBX Server (2000 users, 300
	Total		Rate in Words	Rate	Qty	Unit	em
	ي	प्रतिश	Procurement of IP Telephone and Installation	of IP Telep	urement	Pro	
कि प्रशासकीय आयक्ष		न्यमितिकाका इसक अन्तर	e of Goods	Price Schedule of Goods	Pı		
जाराज निरोला		THE PROPERTY OF THE PARTY OF TH					
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7	6	5	4	G	2		NS		
Installation and Configuration Charge of IP Phone	Installation and Configuration Charge of PBX	SIP Trunk Configuration Charges	Extension Screen module (min 20 programmable dual color buttons, 2 pages per module)	4 lines 4 SIP Accounts POE+ IP Telephone (Must support Extension Screen)	2 lines 2 SIP Account POE IP Telephone, > 2.2 inch Screen (5 way audio conferencing, noise shield technology, 2 RJ 45 10/100 ports)	GIGABIT IP PBX Server (2000 users, 300 concurrent cals, open source perpetual license, support desktop, android app, and SIP Endpoints, Integrated POE/POE+ ports, supports opus voice codec and H.264/H.263/ H.263+/H.265/VP8 video codec.)	Item		
pcs	pcs	lot	pcs	pcs	pcs	pcs	Unit	Procu	
91	-	1	1.	1	90	i	Qty	rement	Pri
							Rate	of IP Teleph	Price Schedule of Goods
							Rate in Words	one and Installation	of Goods
								त्रे प्रदेश,	नेयमितिकाका क
							Total		क्रिक्ट प्रशासक
	Installation and Configuration Charge of IP Phone pcs 91	Installation and Configuration Charge of PBX pcs Installation and Configuration Charge of IP Phone pcs	SIP Trunk Configuration Charges lot Installation and Configuration Charge of PBX pcs Installation and Configuration Charge of IP Phone pcs	Extension Screen module (min 20 programmable dual color buttons, 2 pages per module) SIP Trunk Configuration Charges Installation and Configuration Charge of PBX Installation and Configuration Charge of IP Phone pcs	4 lines 4 SIP Accounts POE+ IP Telephone (Must support Extension Screen) Extension Screen module (min 20 programmable dual color buttons, 2 pages per module) SIP Trunk Configuration Charges Installation and Configuration Charge of PBX pcs Installation and Configuration Charge of IP Phone pcs	2 lines 2 SIP Account POE IP Telephone, > 2.2 inch Screen (5 way audio conferencing, noise shield technology, 2 RJ 45 10/100 ports) 4 lines 4 SIP Accounts POE+ IP Telephone (Must support Extension Screen) Extension Screen module (min 20 programmable dual color buttons, 2 pages per module) SIP Trunk Configuration Charges Installation and Configuration Charge of PBX Installation and Configuration Charge of IP Phone pcs	GIGABIT IP PBX Server (2000 users, 300 concurrent cals, open source perpetual license, support desktop, android app, and SIP Endpoints, Integrated POE/POE+ ports, supports opus voice codec and H.264/H.263/ H.263+/H.265/VP8 video codec.) 2 lines 2 SIP Account POE IP Telephone , > 2.2 inch Screen (5 way audio conferencing, noise shield technology, 2 RJ 45 10/100 ports) 4 lines 4 SIP Accounts POE+ IP Telephone (Must support Extension Screen) Extension Screen module (min 20 programmable dual color buttons, 2 pages per module) SIP Trunk Configuration Charges Installation and Configuration Charge of PBX Installation and Configuration Charge of IP Phone pcs	Item Unit Qty Rate Rate in Words GIGABIT IP BBX Server (2000 users, 300 concurrent cals, open source perpetual license, support desktop, android app, and SIP Endpoints, largerated POE/POE+ ports, supports opus voice codec and H.264/H.263/H.263/H.263+H.265/VP8 1 lines 2 SIP Account POE IP Telephone , > 2.2 inch Screen (5 way audio conferencing, noise shield technology, 2 RJ 45 10/100 ports) 4 lines 4 SIP Accounts POE+ IP Telephone (Must support Extension Screen) Extension Screen module (min 20 programmable dual color buttons, 2 pages per module) SIP Trunk Configuration Charges Installation and Configuration Charge of PBX Installation and Configuration Charge of IP Phone pcs 91 Installation and Configuration Charge of IP Phone pcs 91	Item Unit Qty Rate Rate in Words Total

Name of Firm Seal of Firm

Signature

Note: The Price is inclusive of all taxes, installation cost and transportation Cost upto

Damak Municipality and Ward Offices



Technical Specification of IP PBX Telephone System

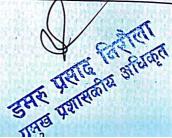
	स्मवः-१
S. N	A property of the property of
1	The Hardware based/server-gateway IP-PBX/PABX/Communication System shall employ IP at its core with IP switching technology and 100% non-blocking based.
2	The system shall provide IP functionality at its core to support SIP/IP extensions and trunks over SIP protocol. It should be possible to support SIP Trunks and SIP/IP Extension with the single VOIP module. It should support minimum 1500 SIP/IP users (SIP/IP Phone, Mobile softphones, UC Client) without any change in Hardware
	System should support SIP Trunking to Internet Telephony Service Providers, allowing non-SIP phones to make SIP calls
3	Operating Platform of PBX should not work on free downloadable software's or Asterisk applications loaded on a PC/Server
4	The system should have capability to handle at least 250 concurrent calls and 200 SRTP concurrent calls.
	System should support mobility, IM and presence, Web-Collaboration, Messaging and centralized licensing.
	The System should have in built-in DHCP Server, which should be able to give IP Addresses to the endpoints.
5	System power supply should be having redundant power input supply. Input: 100~240VAC, 50/60Hz;
	System should support WebRTC based application for Web Collaboration users for Windows and Mac OS
6	The system should have at least two self-adaptive Gigabit ports (switched, routed or dual mode) LAN and WAN to separate out local and VOIP traffic on external network.
7	The System Should support G.711, G.729, G.729a & G.722 or any less utilization bandwidth across WAN and LAN
	The proposed communication system should have a highly secured, encrypted IP supporting hybrid Trunks e.g. Analog CO, IP Trunks (SIP)
9	The system should support following UC features:
	Video Calling with screen sharing
	Audio conferencing
	Fax to email
-1717	Busy Lamp Field
	Automated NAT firewall traversal service facilitates secure remote connections
10	System should support media encryption: SRTP, TLS, HTTPS, SSH, 802.1X and voice codec Opus, G.711 A-law/U-law, G.722, G722.1 G722.1C, G.723.1 5.3K/6.3K, G.726-32, G.729A/B, iLBC, GSM; T.38
12	It should be suitable for DTMF as well as the FSK type of telephone instruments.
13	It should have built-in multi-party video Conference with minimum 50 parties with 1080P with Web conferencing without any software licensing. It should have at least 6 conference rooms
4	It should have built-in multi-party audio Conference with minimum 150 parties without any software from Day-1
5	The System Should support Multiple conferences with variable number of users should be possible within each of the conferencing banks.
6	The System should be able to generate detailed reports about the conference.
7	The System should be able to send emails to all the participants giving them the conferencing details
100000000	The System should support scheduling of conference call.
1112	The state of the s
5	The system shall have the inbuilt Auto-attendant facility
5	

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Design of the later	The system shall have built-in web-based software programming tool for system administration
19	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.
20	The call ringing sequence would be programmable and have options such as simultaneous, ring group, round robin and random.
21	Detail reports of all system parameters should be generated by the system
22	The system should have built-in FXS port which can be used as Power Failure Transfer for minimum 4 ports of PSTN Trunks. No external devices for Power failure should be required.
23 ·	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.
24	Access codes, system timers and access to features shall be programmable.
25	System should support dial from the corporate directory. There shall be minimum 999 numbers possible to store in corporate directory and shall also possible to dial it as an abbreviated number.
26	Features given to an extension shall be accessed from any other extension by dialing the secret codes.
27	The system must have following features:
	Multiple call routes option on Trunk
WHI	CLI based DISA (Mobile Extension), DID, DOD
	Multi-stage Dialing
Train	Multi-layer of IVR in Multiple languages
	Full API for third party Integration
	Dual Ring
	Routing of calls to only permissible legal networks (Logical Partitioning)
	CDR reports and LDAP features
29	Extension features shall have an extension to extension call, extension to central office, extension to operator, call transfer, call forward, speed dial, call back, dial by name, emergency call, call follow me, blacklist/whitelist, call wakeup, do not disturb, barge-in, Priority shall be supported.
80	Operator features shall have the assistance to extension, attended call transfer, indication of call waiting, night service control etc. should be available by default.
31	The system shall have features as Multiple configurable call queues, automatic call distribution
	(ACD) based on agent skills/availability/work-load, in-queue announcement
32	(ACD) based on agent skills/availability/work-load, in-queue announcement The system shall have a conversational recording. Conversation recording should be possible on Analog/IP as well as Mobile SIP Smartphones (Android/iPhone) without any additional software licenses.
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33	The system shall have a conversational recording. Conversation recording should be possible on Analog/IP as well as Mobile SIP Smartphones (Android/iPhone) without any additional software licenses. Varied type of open SIP Terminals such as IP Phone, SIP softphone, Mobile SIP Client and UC Client shall be supported.
3	The system shall have a conversational recording. Conversation recording should be possible on Analog/IP as well as Mobile SIP Smartphones (Android/iPhone) without any additional software licenses. Varied type of open SIP Terminals such as IP Phone, SIP softphone, Mobile SIP Client and UC Client shall be supported. The manufacturer should also have Client application for Android and iPhone and on Windows
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3 4 5	The system shall have a conversational recording. Conversation recording should be possible on Analog/IP as well as Mobile SIP Smartphones (Android/iPhone) without any additional software licenses. Varied type of open SIP Terminals such as IP Phone, SIP softphone, Mobile SIP Client and UC Client shall be supported. The manufacturer should also have Client application for Android and iPhone and on Windows PC so that the mobility can be extended for the Smartphone users. The system must support following features of IP telephony: Dynamic DNS,
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3 4 5	The system shall have a conversational recording. Conversation recording should be possible on Analog/IP as well as Mobile SIP Smartphones (Android/iPhone) without any additional software licenses. Varied type of open SIP Terminals such as IP Phone, SIP softphone, Mobile SIP Client and UC Client shall be supported. The manufacturer should also have Client application for Android and iPhone and on Windows PC so that the mobility can be extended for the Smartphone users. The system must support following features of IP telephony: Dynamic DNS, Registrar Server, Proxy Server, Presence Server, NAT and STUN. The system should support Voice Mail System with following features: Selectively allocate voicemails to users with the flexibility of customizable mailbox size and greetings for All/Selective users



Maria .					
	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				
1813	Notification of a new voicemail via email alert				
37	The product must have certifications like CE, FCC, IC, RCM and ISO.				
38	The proposed IP PBX system should be suitable for future expansion and compatible with an				
40	number of OEMs products/ip phones.				
40	The contractor who is executing the work should be certified and Trained System Integrator of the OEM, and should have valid association / authorization from the OEM for ensuring properties, supply, execution and service support.				
4	Manufacturer Authorization Letter Form is to be provided				
Party.	Wandracturer Numbrization Detter Form is to be provided				
Speci	ification of Operator IP Phone				
5. N	Parameter				
J. 11					
61572	The IP Phone should be Standard VoIP with 4 Line & 4 SIP account				
	The IP phone should have Minimum 4.3-inch LCD display				
	must support Audio Codec: G.723.1, G.729A/B, G.711μ/a, G.726-32, G.722(wide-band), iLBC				
	in-band and out-of-band DTMF (in audio, RFC2833, SIP INFO), HD Audio on handset and				
	Speakerphone"				
	The Phone Must have RJ9 headset port to connect any standards-based headset. The Phone				
7	must also have a separate headset key				
1.74	Corporate Directory and Lightweight Directory access Protocol (LDAP) integration				
V 3	The phone must support minimum 3-way audio Conference				
	The phone must support minimum 200 entries for call history i.e. missed, received, placed etc.				
	The phone must support extension modules				
4.5	Must support busy lamp indicator (BLF) to indicate the presence				
3	The phone must support the following features at a minimum:				
	a. Call forward				
3.6	b. Call pickup				
111	c. Call waiting				
. 1	d. Callback				
	e. Call park				
130	f. Conference				
	a Descend directory				
	g. Personal directory h. Auto answer				
	i. Music on hold (MoH)				
	j. SIP URI dialing				
	k. Message waiting indicator (MWI)				
43	1. Call history lists				
1					
	The phone must have Dual switched 10/100/1000 Mbps ports with integrated PoE, one for				
. (LAN connection and other for connecting to PC/Laptop The phone should have Photoseth V2.1 is built for Photoseth headest support				
	The phone should have Bluetooth V2.1 in-built for Bluetooth headset support.				
	The phone can should have Expansion modules which features at least 20 quick-dial/BLF key which dual-color LED, 2 navigation keys for operator				
1.3	The Phone Must Support server redundancy and fail-over				
4.5	The phone should support auto provisioning				

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	The phone should be from same manufacturer for best Feature Compatibility
	Manufacturer Authorization Letter Form is to be provided
Speci	fication of Basic IP Phone for normal user
S. N	Parameter
1	The phone must have minimum 2 Lines and 2 SIP accounts
	Minimum 2" LCD display
	Must support following audio codec - G.711a, G711u, G.729a, G722, iSAC, Internet Low Bitrate Coded (iLBC)
	The Phone Must have RJ9 headset port to connect any standards-based headset. The Phone must also have a separate headset key
	The phone Must Support downloadable phonebook (XML, LDAP, up to 1000 items)
	The phone must support adjustable desk placement and wall mounting
	The phone must support minimum 200 entries for call history i.e. missed, received, placed etc.
	The phone must support the following features at a minimum:
	a. Call forward
	b. Call pickup
	c. Call waiting
	e. Call park
	f. Conference
	g. Auto answer
	h. Hot desking
3	The phone must have Dual switched 10/100 Mbps ports with integrated PoE, one for LAN connection and other for connecting to PC/Laptop
	The phone must support minimum 3-way audio Conference
	The Phone Must Support server redundancy and fail-over
4	The phone should support auto provisioning
	The phone should be from same manufacturer for best Feature Compatibility
	Manufacturer Authorization Letter Form is to be provided
Specif	fication of Extension Module
S. N	Parameter
~.11	128x384 backlit LCD display
	20 programmable dual-color buttons per module, 2 pages per module (40 contacts total
	BLF/speed dial
	Daisy-chain up to 4 modules for up to 160 contacts/extensions
	BLA (bridged line appearance)/SCA (shared call appearance), BLF (busy lamp field, standard or eventlist), Call Park/Pick-up, Speed Dial, Presence, Intercom, and conference/ transfer/forward and more
	Lain Berner 2017년 2월 2일 전 2018년 1일



