Anexa nr. 22
la Documentația standard nr._____
din "____" ________ 20___

Specificații tehnice

[Acest tabel va fi completat de către ofertant în coloanele 2, 3, 4, 6, 7, iar de către autoritatea contractantă – în coloanele 1, 5,]						
Numărul procedurii de achizițieocds-b3wdp1-MD-1692607827084 din 21 august 2023						
Obiectul achiziției: Echipamentului informational şi de Telecomunicații (inclusiv IP telefonie) pentru anul 2023						
Denumirea bunurilor/serviciilor	**Denumirea modelului bunului/serviciului**	**Țara de origine**	**Produ-cătorul**	**Specificarea tehnică deplină solicitată de către autoritatea contractantă**	**Specificarea tehnică deplină propusă de către ofertant**	**Standarde de referință**
Bunuri						
Lot 10 IP Telefonie						
Lot 10.1 Telefon IP tip 1	GXP1620_1625	ROMANIA	GRANDSTREAM	Telefon IP tip 1 Taste Funcționale: 2 cont SIP, 2 taste de linie cu LED-uri bicolore, 3 taste soft programabile sensibile la context. Afișaj grafic: display LCD de 132 x 48 (2,95 inci). Interfețe de rețea: porturi Ethernet duale 10/100 Mbps, suport Poe Suport protocoale și standarde: SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, NTP, STUN, SIMPLU, LLDPMED, LDAP, TR-069, 802.1x, TLS, SRTP, CDP/SNMP/RTCP-XR Suport codecuri vocale pentru: G.711µ/a, G.722 (bandă largă), G.723,G.726-32, G.729 A/B, iLBC, DTMF în bandă și în afara benzii (În audio, RFC2833, SIP INFO), VAD, CNG, AEC, PLC, AJB, AGC Conținutul pachetului: telefon, receptor cu cablu, suport de bază, alimentare universală, cablu de rețea.	Protocols/Standards SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, NTP, STUN, SIMPLE, LLDP-MED, LDAP, TR-069, 802.1x, TLS, SRTP, CDP/SNMP/RTCP-XR Network Interfaces Dual switched auto-sensing 10/100 Mbps Ethernet ports, integrated PoE (GXP1625 only) Graphic Display 132 x 48 (2.95'') backlit graphical LCD display Feature Keys	

				Garantie 24 luni (cutiile/ambalajul nu se pastrează sau le pastreaza furnizorul) Toate telefoanele trebuie sa fie de la acelasi producator	2 line keys with dual-color LED and 2 SIP accounts, 3 XML programmable context sensitive soft keys, 5 (navigation, menu) keys. 13 dedicated function keys for MUTE, HEADSET, TRANSFER, CONFERENCE, SEND and REDIAL, SPEAKERPHONE, VOLUME, PHONEBOOK, MESSAGE, HOLD, PAGE/INTERCOM, RECORD, HOME Voice Codecs Support for G.711µ/a, G.722 (wide-band), G.723, G.726-32, G.729 A/B, iLBC, in-band and out-of-band DTMF (In audio, RFC2833, SIP INFO), VAD, CNG, AEC, PLC, AJB, AGC Telephony Features Hold, transfer, forward (unconditional/no-answer/busy), 3-way conferencing, call park/pickup, shared-call appearance (SCA) / bridged-line-appearance (BLA), Downloadable phone book (XML, LDAP, up to 1000 items), call waiting, call history (up to 200 records), off-hook auto dial, auto answer, click-to-dial, flexible dial plan, hot desking, personalized music ringtones, server redundancy & fail-over Headset Jack RJ9 headset jack (allowing EHS with Plantronics headsets) HD Audio Yes, HD handset and speakerphone with support for wideband audio Base Stand Yes, 2 angled positions available, wall mountable Wall Mountable	

					Yes QoS Layer 2 QoS (802.1Q, 802.1P) and Layer 3 (ToS, DiffServ, MPLS) QoS Security User and administrator level access control, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, TLS, SRTP, HTTPS, 802.1x media access control Multi-language English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, simplified and traditional Chinese, Korean, Japanese and more Upgrade/Provisioning Firmware upgrade via TFTP / HTTP / HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file, FTP/FTPS Power & Green Energy Efficiency Universal Power Supply Input 100-240VAC 50-60Hz; Output +5VDC, 600mA; PoE: IEEE802.3af Class 2, 3.84W-6.49W; IEEE802.3az (EEE) (GXP1625 only) Physical Dimension: 209mm (L) x 184.5mm (W) x 76.2mm (H) (with handset) Unit weight: 0.73kg; Package weight: 1.1kg Temperature and Humidity Operation: 0°C to 40°C, Storage: -10°C to 60°C , Humidity: 10% to 90% Non-condensing Package Content GXP1620/1625 phone, handset with cord, base stand, universal power supply, network	

					cable, Quick Installation Guide, brochure, GPL License Compliance FCC: Part 15 (CFR 47) Class B CE : EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1 RCM: AS/ACIF S004; AS/NZS CISPR22/24; AS/NZS 60950; AS/NZS 60950.1	
Lot 10.2 Telefon IP tip 2	GXP2170	ROMANIA	GRANDSTREAM	Telefon IP tip 2 Taste caracteristici: 5 conturi SIP, 10 taste de linie. Display grafic: LCD color TFT de 4,3 inci (480x272). Interfață: interfață Wi-Fi dual-band integrată 802.11 a/b/g/n/ac (2,4GHz și 5GHz) și Bluetooth integrat. Interfețe de rețea: porturi dual Gigabit Ethernet, suport Poe Suport protocoale și standarde: SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS(A record, SRV, NAPTR), DHCP, PPPoE, TELNET, TFTP, NTP, STUN, SIMPLE, LLDP, LDAP, TR-069, 802.1x, TLS, SRTP, IPV6 Suport codecuri vocale pentru: G.711µ/a, G.722 (bandă largă), G.723,G.726-32, G.729 A/B, iLBC, DTMF în bandă și în afara benzii (În audio, RFC2833, SIP INFO), VAD, CNG, AEC, PLC, AJB, AGC Modul de extensie suport 1 buc cu 20 de taste de extindere cu două culori și 2 taste săgeți pentru comutarea paginii. Alimentare: Adaptor universal de alimentare inclus: Intrare: 100 - 240V; Power-over-Ethernet integrat (802.3af) Conținutul pachetului: telefon, receptor cu cablu,suport de bază, alimentare universală, cablu de rețea. Garantie 24 luni (cutiile/ambalajul nu se pastrează sau le pastreaza furnizorul) Toate telefoanele trebuie sa fie de la acelasi producator	Protocols/Standards SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TELNET, TFTP, NTP, STUN, SIMPLE, LLDP, LDAP, TR-069, 802.1x, TLS, SRTP, IPV6, CDP/SNMP/RTCP-XR Network Interfaces Dual switched auto-sensing 10/100/1000 Mbps Gigabit Ethernet ports with integrated PoE Graphic Display 4.3 inch (480x272) TFT color LCD Bluetooth Yes, integrated Feature Keys 12 line keys with up to 6 SIP accounts, 5 XML programmable context sensitive softkeys, 5 navigation/menu keys, 11 dedicated function keys for : MESSAGE (with LED indicator), PHONEBOOK, TRANSFER, CONFERENCE, HOLD, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOL+, VOL- Voice Codecs Support for G7.29A/B, G.711µ/a-law, G.726, G.722(wide-band), G723.1, iLBC, Opus, in-band and out-of-band DTMF(in	

					audio, RFC2833, SIP INFO), VAD, CNG, AEC, PLC, AJB, AGC Auxiliary Ports RJ9 headset jack (allowing EHS with Plantronics headsets), USB, extension module port Telephony Features Hold, transfer, forward, 5-way conference, call park, call pickup, shared-call-appearance (SCA)/bridged-line-appearance(BLA), downloadable phonebook (XML, LDAP, up to 2000 items), call waiting, call log(up to 500 records), XML customization of screen, off-hook auto dial, auto answer, click-to-dial, flexible dial plan, hot desking, personalized music ringtones and music on hold, server redundancy and fail-over HD Audio Yes, HD handset and speakerphone with support for wideband audio Extension Module Yes, can power up to 4 GXP2200EXT modules which features a 128x384 graphic LCD, 20 quick-dial/BLF keys which dual-color LED, 2 navigation keys, and less than 1.2W power consumption per unit. Base Stand Yes, 2 angle positions available, Wall Mountable QoS Layer 2 QoS (802.1Q, 802.1P) and Layer 3 (ToS, DiffServ, MPLS) QoS Security User and administrator level passwords, MD5 and MD5-sess based authentication,	

					256-bit AES encrypted configuration file, SRTP, TLS, 802.1x media access control Multi-language English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, simplified and traditional Chinese, Korean, Japanese and more Upgrade/Provisioning Firmware upgrade via TFTP / HTTP / HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file, FTP/FTPS Power & Green Energy Efficiency Universal power adapter included: Input:100-240V ; Output: +12V, 1.0A ; Integrated Power-over-Ethernet (802.3af) Max power consumption : 5.4W(without GXP2200EXT) or 9.2W(with 4 cascaded GXP2200EXTs) Physical Dimension: 228mm(W) x 206mm(L) x 46mm(H); Unit weight:0.98kg ; Package weight:1.43kg Temperature and Humidity Operation: 0°C to 40°C Storage: -10°C to 60°C Humidity: 10% to 90% Non-condensing Package Content GXP2170 phone, handset with cord, base stand, universal power supply, network cable, Quick Installation Guide, GPL license Compliance FCC: Part 15 (CFR 47) Class B CE: EN55022 Class B; EN55024 Class B;EN61000-3-2; EN61000-3-3;EN60950-1	

					RCM: AS/ACIF S004; AS/NZS CISPR22/24; AS/NZS 60950.1	
Lot 10.3 Telefon IP tip 3	GXP2170 GXP2200ext	ROMANIA	GRANDSTREAM	Telefon IP tip 3 Suportă 6 de linie, 6 conturi SIP și conferințe vocale în 5 căi, 24 taste de apelare rapidă/extensie BLF cu afișaj color dublu, 5 taste programabile sensibile la context, 5 taste de navigare/meniu, 11 taste funcționale dedicate pentru funcții precum: MESAJ AGENDA DE TELEFON, TRANSFER, CONFERINȚĂ, APRIRE APEL, SETUL CĂȚI, MUTE, TRIMITE/REDAPARE, DIBUTOR, VOLUM +, VOLUM - Port de rețea dublu de 10/100/1000 Mbps cu detectare automată și PoE încorporat Display grafic: LCD color TFT de 4,3 inci (480x272) Bluetooth integrat. Interfețe de rețea: porturi dual Gigabit Ethernet, suport Poe Suport protocoale și standarde: SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS(A record, SRV, NAPTR), DHCP, PPPoE, TELNET, TFTP, NTP, STUN, SIMPLE, LLDP, LDAP, TR-069, 802.1x, TLS, SRTP, IPV6 Suport codecuri vocale pentru:G.729A/B, G.711µ/a-law, G.726, G.722 (bandă largă) și iLBC, DTMF în bandă și în afara benzii (în audio, RFC2833, SIP INFO) Alimentare: Adaptor universal de alimentare inclus: Intrare: 100 - 240V; Power-over-Ethernet integrat (802.3af), Consum maxim de energie 12 W sau 12,95 W (PoE) Conținutul pachetului: telefon, receptor cu cablu, Garantie 24 luni (cutiile/ambalajul nu se pastrează sau le pastreaza furnizorul) Telefoanele trebuie sa fie de la acelasi producator	Protocols/Standards SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TELNET, TFTP, NTP, STUN, SIMPLE, LLDP, LDAP, TR-069, 802.1x, TLS, SRTP, IPV6, CDP/SNMP/RTCP-XR Network Interfaces Dual switched auto-sensing 10/100/1000 Mbps Gigabit Ethernet ports with integrated PoE Graphic Display 4.3 inch (480x272) TFT color LCD Bluetooth Yes, integrated Feature Keys 12 line keys with up to 6 SIP accounts, 5 XML programmable context sensitive softkeys, 5 navigation/menu keys, 11 dedicated function keys for : MESSAGE (with LED indicator), PHONEBOOK, TRANSFER, CONFERENCE, HOLD, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOL+, VOL- Voice Codecs Support for G7.29A/B, G.711µ/a-law, G.726, G.722(wide-band), G723.1, iLBC, Opus, in-band and out-of-band DTMF(in audio, RFC2833, SIP INFO), VAD, CNG, AEC, PLC, AJB, AGC Auxiliary Ports	

					RJ9 headset jack (allowing EHS with Plantronics headsets), USB, extension module port Telephony Features Hold, transfer, forward, 5-way conference, call park, call pickup, shared-call-appearance (SCA)/bridged-line-appearance(BLA), downloadable phonebook (XML, LDAP, up to 2000 items), call waiting, call log(up to 500 records), XML customization of screen, off-hook auto dial, auto answer, click-to-dial, flexible dial plan, hot desking, personalized music ringtones and music on hold, server redundancy and fail-over HD Audio Yes, HD handset and speakerphone with support for wideband audio Extension Module Yes, can power up to 4 GXP2200EXT modules which features a 128x384 graphic LCD, 20 quick-dial/BLF keys which dual-color LED, 2 navigation keys, and less than 1.2W power consumption per unit. Base Stand Yes, 2 angle positions available, Wall Mountable QoS Layer 2 QoS (802.1Q, 802.1P) and Layer 3 (ToS, DiffServ, MPLS) QoS Security User and administrator level passwords, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, SRTP, TLS, 802.1x media access control Multi-language	

						English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, simplified and traditional Chinese, Korean, Japanese and more Upgrade/Provisioning Firmware upgrade via TFTP / HTTP / HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file, FTP/FTPS Power & Green Energy Efficiency Universal power adapter included: Input:100-240V ; Output: +12V, 1.0A ; Integrated Power-over-Ethernet (802.3af) Max power consumption : 5.4W(without GXP2200EXT) or 9.2W(with 4 cascaded GXP2200EXTs) Physical Dimension: 228mm(W) x 206mm(L) x 46mm(H); Unit weight:0.98kg ; Package weight:1.43kg Temperature and Humidity Operation: 0°C to 40°C Storage: -10°C to 60°C Humidity: 10% to 90% Non-condensing Package Content GXP2170 phone, handset with cord, base stand, universal power supply, network cable, Quick Installation Guide, GPL license Compliance FCC: Part 15 (CFR 47) Class B CE: EN55022 Class B; EN55024 Class B;EN61000-3-2; EN61000-3-3;EN60950-1 RCM: AS/ACIF S004; AS/NZS CISPR22/24; AS/NZS 60950.1	

					Lines -20 per page (each module contains 2 pages, for up to 40 lines per module -Up to 160 with 4 daisy-chained modules Compatible Grandstream IP phones GXP2140, GXP2170 and GXV3240 Feature Support Local GUI with animation driven from the host GXP2140 or GXP3240 phone; Multiple line/call appearances Power Powered by the host phone Firmware Upgrades Delivered by the host phone Dimensions (L x W x H) 206 mm x 117mm x 32mm Weight 0.38kg Temperature 0 ~ 40ºC (32 ~ 104ºF) Humidity 10%-90% Non-condensing	
Lot 10.4 Statie IP PBX	UCM6302	ROMA NIA	GRAN D STRE AM	Capacitate de apel: Utilizatori: 1000 Apeluri simultane (G.711): 150 Apeluri SRTP simultane maxime (G.711): 100 Interfețe de rețea Trei porturi Gigabit Porturi FXS pentru telefon analogic: 2 rj11 Linie PSTN porturi FXO: 2 rj11 Numărul maxim de participanți la podurile de conferință: 6 săli de conferințe video și până la 30 de petreceri cu 1080p, presupunând 4 fluxuri video + 1 partajare a ecranului (H.264 și Opus) Conferință vocală: până la 150 de persoane (G.711) Codecuri video: H.264, H.263, H263+, H.265, VP8 Codecuri de voce și fax 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 Display LCD 320x240 LCD color cu ecran tactil QoS Layer 2 QoS (802.1Q, 802.1p) și Layer 3 (ToS, DiffServ, MPLS) QoS Protocoale de rețea: SIP, TCP/UDP/IP, RTP/RTCP, IAX, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, STUN, SRTP, TLS, LDAP, HDLC, HDLC -ETH, PPP, Frame Relay (în așteptare), IPv6, OpenVPN® Metode de deconectare: Ton de ocupat/aglomerație/Tonul urlet, inversarea	Analog Telephone FXS Ports 2 RJ11 Ports All ports have lifeline capability in case of power outage; number of ports can be expanded by peering with an FXS gateway PSTN Line FXO Ports 2 RJ11 Ports Network Interfaces Three self-adaptive Gigabit ports (switched, routed or dual mode) with PoE+ NAT Router Yes (supports router mode and switch mode) Peripheral Ports 1*USB 3.0, 1*SD card interface 1*USB 2.0, 1*USB 3.0, 1*SD card interface 2*USB 3.0, 1*SD card interface LED Indicators None	

				polarității, sincronizarea blițului cu cârlig, deconectarea curentului în buclă Criptare media: SRTP, TLS, HTTPS, SSH, 802.1X Intrare sursă de alimentare universală: 100 ~ 240VAC, 50/60Hz; Garantie 24 luni (cutiile/ambalajul nu se pastrează sau le pastreaza furnizorul) Statie IP PBX trebuie sa fie de la acelasi producator ca si telefoanele din acelasi lot	Power 1/2, FXS, FXO, LAN, WAN, Heartbeat LCD Display 320x240 color LCD with touch screen for Shortcut Keys and Scroll Bar 128x32 dot matrix graphic LCD with DOWN and OK buttons Reset Switch Yes, long press for factory reset and short press for reboot Voice-over-Packet Capabilities LEC with NLP Packetized Voice Protocol Unit, 128ms-tail-length carrier grade Line Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711, NetEQ, FEC 2.0, jitter resilience up to 50% audio packet loss Voice and Fax Codecs 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 Video Codecs H.264, H.263, H263+, VP8 QoS Layer 2 QoS (802.1Q, 802.1p) and Layer 3 (ToS, DiffServ, MPLS) QoS API Full API available for third-party platform and application integration Telephony Operating System Based on Asterisk version 16 DTMF Methods In-band audio, RFC2833, and SIP INFO Provisioning Protocol & Plug-and-Play	

						Mass provisioning using AES encrypted XML configuration file, auto-discovery & auto-provisioning of Grandstream IP endpoints via ZeroConfig (DHCP Option 66 multicast SIP SUBSCRIBE mDNS), eventlist between local and remote trunk Network Protocols SIP, TCP/UDP/IP, RTP/RTCP, IAX, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, STUN, SRTP, TLS, LDAP, HDLC, HDLC-ETH, PPP, Frame Relay (pending), IPv6, OpenVPN® Disconnect Methods Busy/Congestion/Howl Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect Media Encryption SRTP, TLS, HTTPS, SSH, 802.1X Universal Power Supply Input: 100 ~ 240VAC, 50/60Hz; Output: DC 12V, 1.5A Temperature & Humidity Operating: 32 - 113ºF / 0 ~ 45ºC, Humidity 10 - 90% (non-condensing) Storage: 14 - 140ºF / -10 ~ 60ºC, Humidity 10 - 90% (non-condensing) Mounting Wall mount & Desktop Rack mount & Desktop Multi-Language Support -Web UI: English, Simplified Chinese, Traditional Chinese, Spanish, French, Portuguese, German, Russian, Italian, Polish, Czech, Turkish	

					-Customizable IVR/voice prompts: English, Chinese, British English, German, Spanish, Greek, French, Italian, Dutch, Polish, Portuguese, Russian, Swedish, Turkish, Hebrew, Arabic, Nederlands -Customizable language pack to support any other languages Caller ID Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 – BT, NTT Polarity Reversal/Wink Yes, with enable/disable option upon call establishment and termination Call Center Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability/ work-load, in-queue announcement Customizable Auto Attendant Up to 5 layers of IVR (Interactive Voice Response) in multiple languages Wave App Free; Available for desktop (Windows 10+, Mac OS 10+), web (Firefox and Chrome Browsers) and mobile (Android & iOS), allows users to join UCM-hosted meetings/conferences, communicate with other users/solutions and make/receive calls using SIP accounts registered to a UCM6300 series IP PBX Call Features Call park, call forward, call transfer, call waiting, caller ID, call record, call history, ringtone, IVR, music on hold, call routes, DID, DOD, DND, DISA, ring group, ring simultaneously, time schedule, PIN groups, call queue, pickup group, paging/intercom,	

					voicemail, call wakeup, SCA, BLF, voicemail to email, fax to email, speed dial, call back, dial by name, emergency call, call follow me, blacklist/whitelist, voice conference, video conference, eventlist, feature codes, busy camp-on/ call completion, voice control, post-meeting reports, virtual fax sending/receiving, email to fax Firmware Upgrade Supported by Grandstream Device Management System (GDMS), a zero-touch cloud provisioning and management system, It provides a centralized interface to provision, manage, monitor and troubleshoot Grandstream products Internet Protocol Standards RFC 3261, RFC 3262, RFC 3263, RFC 3264, RFC 3515, RFC 3311, RFC 4028. RFC 2976, RFC 3842, RFC 3892, RFC 3428, RFC 4733, RFC 4566, RFC 2617, RFC 3856, RFC 3711, RFC 4582, RFC 4583, RFC 5245, RFC 5389, RFC 5766, RFC 6347, RFC 6455, RFC 8860, RFC 4734, RFC 3665, RFC 3323, RFC 3550 Compliance FCC: Part 15 (CFR 47) Class B, Part 68 CE: EN 55032, EN 55035, EN 61000-3-2, EN 61000-3-3, EN 62368-1, ETSI ES 203 021, ITU-T K.21 IC: ICES-003, CS-03 Part I Issue 9 RCM: AS/NZS CISPR 32, AS/NZS 62368.1, AS/CA S002, AS/CA S003.1/.2 Power adapter: UL 60950-1 or UL 62368-1	

					6 Video Conference rooms and up to 30 parties with 1080p, assuming 4 video feeds + 1 screen sharing (H.264 & Opus) Voice Conference: Up to 150 parties (G.711)	
Lot 10.5 Servicii de instalare si configurare telefonie IP				Servicii de instalare si configurare telefonie IP (Configurare Statie, telefoane, IVR, conectarea la serviciul IP telefonie oferit de furnizor) inslusiv instruirea personalului reponsabil	Servicii de instalare si configurare telefonie IP (Configurare Statie, telefoane, IVR, conectarea la serviciul IP telefonie oferit de furnizor) inslusiv instruirea personalului reponsabil	

Numele, Prenumele: Vitalie Bîrsan În calitate de: Administrator
Ofertantul: ICS Reliable Solutions Distributor SRL Adresa: Mun. Chișinău str. Alexandru cel Bun 85

Digitally signed by Bîrsan Vitalie
Date: 2023.11.03 10:24:12 EET
Reason: MoldSign Signature
Location: Moldova