SPECIFICATIONS FOR A UNIFIED COMMUNICATION SYSTEM / VOICE OVER INTERNET PROTOCOL (VOIP) SOLUTION
1.0 PARTICULARS

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1.1 Scope of Work

The scope shall cover, detailed site survey, design, supply, delivery, installation, configuration, testing and commissioning of a Unified Communication System (VOIP phones). The system shall be installed in Kisumu new Judiciary building.

The new system shall be linked to the existing system in the old building where the call manager system is located.

<table>
<thead>
<tr>
<th>Item</th>
<th>Standard IP phones</th>
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<tbody>
<tr>
<td>1</td>
<td>Cisco 7945G/7942G</td>
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<td>2</td>
<td>50</td>
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2.0 SYSTEM DESCRIPTION AND COMPOSITION

The Internet Protocol (IP) PBX System shall be a complete telephony system that will provide telephone calls over IP data networks. All conversations shall be sent as data packets over the network. The technology shall include advanced communication features but will also provide significant scalability and robustness. The IP PBX shall also be able to connect to TELKOM/SAFARICOM/AIRTEL/ORANGE/YU lines via a PSTN/GSM/E1 trunk interface. The following is the description of the major components of the proposed IP PBX system.

2.1 Primary Services

a. Centralized Call Processing

b. The IP PBX system shall be able to support initially 100 IP users with expansion capacity of up to 200.

c. The IP PBX system shall be able to support 20 trunk lines.
d. The system shall be able to support full outlook and e-mail integration including voice mail to Email.

e. The system shall have support for business critical applications and unified communication applications which shall include IP Contact Centre/Call Centre, unified messaging.

f. The system shall have support for integrated services such as conferencing, one number follow me, personal call directory, recorded announcement, network-wide attendant and messaging.

g. The system shall have an IP PBX based unified communication security solutions that offer comprehensive threat protection, strict policy enforcement, robust access control, and privacy of confidential data.

h. The system shall be able to interoperate with IP Phones, PSTN Gateways, SIP Trunks, IP Trunks, Analogue Trunks and Microsoft products such as Exchange 2010 or lower versions. The system shall have a CTI server to be able to provide call management on every desktop. The system shall make it possible for the following feature to be integrated:

   - **Pop-up of incoming calls with caller ID on ISDN line, List of incoming/outgoing and answered/unanswered calls.**
   - **Click to dial.**
   - **Integration with MS outlook to dial from Outlook contacts.**
   - **Support in future, unified messaging to have Voicemail as .wav file in email client.**
   - **Support in future, supervision feature to have status Free/Busy/forwarded etc.) of all users.**

i. The system shall have multi-party conference with clock to conference and multiparty video conferencing.

j. The system shall be flexible enough to allow KSB configure any preferred numbering plan, including capability to support all extension on Direct Inward Dialing (DID) numbering.
2.2 Secondary Services

I. Cell Phone Integration
II. Extension Groups
III. Time of Day Routing
IV. Extension Call Recording
V. Extension Range Flexibility
VI. Voicemail Bypass
VII. Intuitive VoIP Ready
VIII. Announcement Interface

2.3 Phone Sets

The system shall be able to support extensive user equipment including cordless, soft phones, IP, digital and analogue phone set support to meet diverse end-user requirement. Specifically the system shall be able to support the following; Digital Sets, Analogue Sets, IP phones (IP Key phones / Soft phones), SIP Phones (SIP desk phones / soft phones), DECT Phones and V-WLAN (WIFI Sets).

i. The Digital/IP sets shall have large display (minimum of 3.7 inches x 2.1 inches or 9.5 cm x 5.4 cm) and permanently-labeled feature buttons: eg. Speaker, Mute, Volume, Headset, Contacts, Home, History, Message, Phone

ii. The Supplier shall supply IP Phone sets capable of being powered through Power over Ethernet (PoE). The Supplier shall also indicate, as an option IP Phones powered through Power-over-LAN switch.

iii. The Supplier shall supply the specified headsets as indicated in the table above.

iv. The Supplier shall supply client software for the IP soft phones to be installed on desktop and laptop computers. The Supplier shall also supply all associated accessories for each IP soft phones that is: - mouth pieces, mikes, earpieces, headphones and holders.

2.4 Operations and Management Console Software

i. The system shall have a secure Web based IP PBX management system for easy system administration. It shall also be password protected and accessible over the network.

ii. Multiple Country number selection.

iii. Talk Time Management
iv. Call Account Management  
v. Call Detail Records - Call Logs  

2.5 Survey  
Depending on the architecture and configuration of the IP PBX arising from the network design submitted by contractor the supplier will be expected to carry a comprehensive site survey to establish the exact number of PoE port density requirements or any other requirements.

2.6. Automated Call Distributor  
The system shall be able to support an automated call distributor with the following features;  
i. Automatic call attendant  
ii. Call menu  
iii. Call forwarding  
iv. Call transfer  
v. Managing extensions  
vi. Call parking  
vii. Support auto attendant (16 ports) with the option to upgrade it to multiple trees, by way of software upgrade only with no additional hardware.

2.9 Unified Messaging Gateway  
The system shall have a Unified Messaging Gateway which shall be used to link with Microsoft Exchange
3.0 General Specifications

VOIP equipment shall employ adopt commonly used specifications which include;

Area

General

- H.323 or the Session Initiation Protocol (SIP) signaling protocols that sets up, maintain and terminate a VoIP call.
- Media Gateway Control Protocol (MGCP) that provides a signaling and control protocol between VoIP gateways and traditional PSTN (Public Switched Telephone Network) gateways.

Security

- H.323 protocol is secured by using TLS and S/MIME encryption for SIP.
- Adequate physical security is in place to restrict access to key VoIP servers and components.
- Firewalls designed for VOIP protocols are employed to secure the VOIP systems.
- VOIP Terminals are secured through password authentication and user authorization. User accounts shall be administered and managed by the ICT units.
- Wi-Fi Protected Access (WPA) where mobile units are to be integrated with the VOIP system.
- Disabling of HTTP and Telnet services
• Where softphones are used, PCs should be adequately secured to protect from worms, viruses, and other malicious software.
• Creating awareness to users on how to secure use VOIP systems.

Protocols

• Real-Time Transport Protocol.
• Session Initiation Protocol.
• ITU-T H.323
• Media Gateway control protocol
• IPsec, TLS and S/MIME for encryption.

Other Provisions

i. Traditional calling features including call by name, caller ID, last number redial, hold, call waiting, call forwarding, transfer, divert, park, retrieve, voice mail, return call and call conferencing

ii. Call Coverage Make it easy to ensure that important calls are answered by administrative assistants or team members, via user-controlled Delegation and Team Calling respectively.

iii. Telephone Directory.

iv. Maintain Call history.

v. Local Number portability, that is, ability to maintain phone numbers when one changes service provides.