

**PEMASANGAN SISTEM TEKNOLOGI MAKLUMAT DAN KOMUNIKASI (ICT) UNTUK
UNIVERSITI MALAYA – KOMPLEKS PUSAT ASASI SAINS**

TECHNICAL SPECIFICATION – PABX SYSTEM

1. GENERAL SPECIFICATIONS

1.1. SCOPE

1.1.1. This document describes the design requirement in relation to the installation, commissioning and testing the PABX and Telephone equipments for the proposed *Kompleks Pusat Asasi sains Universiti Malaya* in Kuala Lumpur.

1.2. Initial system configuration

1.2.1. The initial capacity requirements are defined in the Bill of Quantity which is to telephone services for 114 extension lines and 4 direct lines respectively.

1.2.2. The PABX telephone system shall have an upgrade path to cater for future IP Telephony .

1.3. Technical requirements

1.3.1. The PABX system the associated equipment shall be of digital type complying with ISDN requirements and provided with all the latest facilities and features such as call transfer, intercommunication, abbreviated dialing, direct inward dialing, detail call recording with print out, on-hold music, paging facilities, etc.

1.3.2. The PABX system shall be of the expandable type. The PABX and the associated equipment shall have sufficient capacity for the present requirement and at least 20% spare capacity for future expansion. All terminal blocks for external cables whether trunk lines or extension lines shall be provided with properly rated lightning arrestors. The PABX shall also be provided with built-in surge suppressor device complying to MS IEC 61643-21 (SPD Connected To Telecommunication & Signalling Network - Performance Requirement & Testing Methods)

1.3.3. The battery capacity shall be not less than one hour at the rated capacity of the PABX, including future extension. The battery shall be maintenance free

type installed in the PABX room or in a separate room adjacent to the UPS equipment. The PABX rooms and the battery rooms shall be 24 hours air-conditioned.

- 1.3.4. The installation shall be in accordance with the guidelines of The Telecommunication Authority. Sufficient number of telephone outlet terminals shall be provided for the present usage and for future extension. Digital telephone sets suitable for both voice and data communication shall be installed where appropriate. The project shall be provided with sufficient number of telephone outlet.
- 1.3.5. The system shall have a distributed Architecture wherein each Gateway shall have a Functionality to function on its own and when all Gateways are connected the various Hardware and Software units shall form a part of the same Distributed communication system.
- 1.3.6. Remote location should be able to function independently even in case of Link Failure with Main site. (Meaning local survivability should be available).
- 1.3.7. The system software should be modular, i.e. divided into separate function blocks which can be added or removed without disturbing the other functional blocks.
- 1.3.8. The offered system must be expandable to support a capacity of 30,000 ports with all media gateway having the same system software versions as part of one distributed system, forming a common numbering plan.
- 1.3.9. The system must be capable to support minimum of 10000 Extensions spread across multiple locations to cater future requirements of UM Campus
- 1.3.10. The system should have the capacity, to support minimum of 100 locally survivable Media gateways as part of the same distributed Architecture.
- 1.3.11. System Must Support up to 250 Operators in a distributed architecture.
- 1.3.12. System must have inbuilt 8 Party conferencing.
- 1.3.13. The system should support the following trunk interfaces:
 - ISDN PRI Trunks from Public Exchange- PSTN as well as PLMN (Public Land Mobile Network).

- H.323 Trunks for IP Connectivity.
- Analog trunk interface from Public exchange.
- Fixed cellular terminals interfaced over analog trunks.

The system must be able to handle following digital tie line interfaces:

- DPNSS
- 1SDN/QS1G

- 1.3.14. System shall support Analogue extensions with a capacity of 32 extensions per Board in the system cabinet
- 1.3.15. System shall support Digital Extensions with a capacity of 32 extensions per Board in the system cabinet.
- 1.3.16. System shall support DECT Cordless extensions with a Capacity to connect up to 8 DECT Base stations and up to 32 DECT Channels per DECT Board. In the system cabinet.
- 1.3.17. IP extensions Boards supporting up to 32 Gateway and 32 Non Gateway channels.
- 1.3.18. Integrated Mobile extensions with DTMF sender and Receiver Board- it shall be possible to Have Users with only Mobile Phones as extensions without needing any other Physical Phones.
- 1.3.19. It is desired to have CLI facility on Analog Extension for calls landing through ISDN PRI Trunks. Bidder must quote accordingly.
- 1.3.20. The SYSTEM should support the facility of at least 1000 abbreviated numbers for common use.
- 1.3.21. The SYSTEM should .support both the hotline and delayed hotline facility.
- 1.3.22. The SYSTEM should support the use of account codes for alternative charging.
- 1.3.23. The SYSTEM should support Emergency extensions at which any seven callers can call in simultaneously and automatically become the part of conference for emergency discussion.

- 1.3.24. The SYSTEM should support generic extensions with an extension number but without any physical phone connected to it, thereby the same can be utilized, such extensions shall have a number, name and generic service profile and the user shall be able to login from any physical phone.
- 1.3.25. The SYSTEM should support Alternative Routing wherein a trunk route must support at least 3 alternate routes.
- 1.3.26. The SYSTEM should support class of service codes to restrict or allow each extension certain type of calls and/or facilities.
- 1.3.27. The SYSTEM should support authorization codes to be dialed from any extension in order to get a higher class of service.
- 1.3.28. The SYSTEM should support Direct Inward System Access (DISA) allowing external users to get access to the SYSTEM features.
- 1.3.29. It should be possible to switch the SYSTEM into the emergency state allowing only some extensions to make external calls.

1.4. System supervision programs

The software shall contain the following options:

- 1.4.1. The 1/0 programs shall keep the following functions:
- 1.4.2. Log on/tog off, user id and passwords for authorization, printout Administration
- 1.4.3. Management and recovery functions
- 1.4.4. The SYSTEM should support an integrated interface to the LAN based on TCP/IP and SNMP for management purposes. The IT administrator shall be able to administer the system from anywhere on the TCP/I-P network.

1.5. User Facilities

- 1.5.1. The SYSTEM should support both Common and Individual Abbreviated Dialing.

1.5.2. The SYSTEM should support Automatic Call Back on:

- a busy extension
- no reply
- busy outgoing lines

1.5.3. The SYSTEM must support Repeated internal Diversion for all users with up to 10 user defined answering Positions.

1.5.4. The SYSTEM should support Call diversion on;

- no reply
- busy

1.5.5. The SYSTEM should support both internal and external follow-me feature.

1.5.6. The SYSTEM should support Call Pick-up for

- common bell group
- answering group
- individual call

1.5.7. The SYSTEM must support feature of free seating which means that it should be possible for the user to logon to any available telephone handset and get the users personal categories, calls and messages.

1.5.8. The SYSTEM should support conference of up to 8 parties.

1.5.9. The SYSTEM should support Do Not Disturb (individual and group

1.5.10. The extension must be able to transfer a call to another extension or to the operator.

1.5.11. Single Directory number (Personal Number) to enhance user reach ability (5 different profile means 5 different Call flows tike one for office hours, second for non-office hours, third for weekends / Vacations etc). This call flow shall provide up to 10 answering positions (Self Ext. Number, Secretary Number, Voice Mail , Res Number, Mobile Number etc).

1.5.12. The SYSTEM should support parallel ringing for up to three answering position defined for user personal number profile.

- 1.5.13. The SYSTEM shall support inquiry calls with shuffle between both parties.
- 1.5.14. The SYSTEM should support Last External Number Re-dialing
- 1.5.15. The SYSTEM should support parking of a call with individual Call pick-up from any telephone.
- 1.5.16. The SYSTEM should support both direct outgoing calls and outgoing calls via the operator.
- 1.5.17. The SYSTEM should support Direct in-Dialing
- 1.5.18. The SYSTEM should support direct-in lines for direct connection to a predetermined destination
- 1.5.19. The SYSTEM should support direct routing to the extension and routing through the operator for incoming automatic inter-SYSTEM calls.
- 1.4.21 The SYSTEM should support Least Cost Routing.
- 1.5.20. The SYSTEM shall support transit traffic by direct switching as well as by the operator.
- 1.5.21. The SYSTEM shall support trunk call discrimination to block some extensions from calling certain parts of the public network as well as the user network and also numbers in the own exchange.
- 1.5.22. The SYSTEM shall support cost effective ways of networking, like building dynamic leased lines to the destination of choice using operator networks.
- 1.5.23. In case of power failure, the SYSTEM shall be able to automatically connect analogue trunk lines to predetermined extensions.

1.6. Group hunting

- 1.6.1. Incoming calls shall be routed to a free extension according to:
 - sequential hunting
 - evenly distribution (cyclic)
- 1.6.2. All group members shall keep their own number and class of service.
- 1.6.3. An extension shall be able to withdraw from a group.

1.7. Night Service

1.7.1. The night service shall be able to be activated when:

- ad operators are absent
- the procedure is activated from an operator console
- Incoming calls are not answered within a certain time.
- a predetermined time

1.7.2. The SYSTEM shall support the following types of night service:

- Universal night service-Signaled on a common alerting system and answered from any extension by dialing a code
- Common night service-Answered from one common answering position for ail incoming calls.
- Individual night service- Answered by a predetermined position per line. The answering positions shall be programmable on a day-by-day basis.

1.7.3. The answering positions may be an-extension, an extension group or a call pick-up group.

1.7.4. Transfer before answer shall be possible at common- and individual night service.

1.8. Networking

1.8.1. The SYSTEM shall be able to constitute a fully integrated network with remote units without losing any functionality

1.8.2. It shall be possible to support small branches (from about 10 extensions) connected to the network without loss of functionality.

1.8.3. The SYSTEM shall be able to network with other Vendor Systems via PRI/IP Trucking.

1.8.4. It shall be possible to use several transmission media for a tie line connection with full support for network transparency and QSIG. The options for media shall at least include; leased lines, PSTN/ISDN, IP.

- 1.8.5. It shall be possible to dynamically set up a semi permanent connection through the public IDN/ISDN network for use as a tie line route with or without any requirement for signaling on the B-channel.
- 1.8.6. Alternative routing shall provide at least three alternative routes when the ordinary route is fully occupied.
- 1.8.7. The routing mechanism shall ensure that the most economical path is always used. In case a more economical path is freed up during a conversation, transfer of the current call to this path shall be automatic.
- 1.8.8. Overflow to the switched network shall be automatic in case all direct links are busy
- 1.8.9. The IP networking shall be based on standard H.323
- 1.8.10. All network elements involved shall support mechanism for QoS. Tenderers to specify the standard supported.
- 1.8.11. The IP network shall support "direct media routing" and Multi Media Support.
- 1.8.12. The following ISO QSIG features must be supported over the IP network-Call Forwarding, Call Transfer, and Conference, centralized Voicemail, centralized operator etc.
- 1.8.13. The following voice codec shall be supported.
 - G.729 Annex A
 - G.729 Annex A with annex B
 - G.723.1
 - G-711
- 1.8.14. IP Boards should support Inband DTMF detection for IP Networking.
- 1.8.15. System must support H.323 Fast connect procedure.

1.9. Basic Digital Phones must support the following features:

- 1.9.1. 2x24 alphanumeric characters display (Two Line and 24 Character)
- 1.9.2. Volume control Keys-Increase and Decrease

- 1.9.3. Programmable ringer, loudness and tone character
- 1.9.4. Come with 14 Feature keys.- Inclusive of Mute, Hands free and Headset function keys Enquiry key, Transfer key etc.
- 1.9.5. Message waiting indicator LED
- 1.9.6. Integrated Headset port in addition to the Handset port.
- 1.9.7. Adjustable foot console
- 1.9.8. Acoustic chock protection
- 1.9.9. The display must show at least the following information:
- 1.9.10. Time and date
- 1.9.11. Own telephone number
- 1.9.12. Telephone status
- 1.9.13. Incoming/outgoing calls

1.10. Advanced Digital Phones must support the following features:

- 1.10.1. Tilttable Display
- 1.10.2. 3x24 alphanumeric characters display (3 Line and 24 Character).
- 1.10.3. Volume control Keys-, Increase and Decrease
- 1.10.4. Programmable ringer, loudness and tone character
- 1.10.5. 18 Feature keys - Inclusive of 4 Dynamic Soft Keys, Mute, Hands free and Headset function keys .Enquiry key, Transfer key etc.
- 1.10.6. Message waiting indicator LED
- 1.10.7. Integrated Headset port in addition to the Handset port.
- 1.10.8. Adjustable foot console
- 1.10.9. Acoustic chock protection
- 1.10.10. Operator Console features:

The Operator console shall be GUI Based on a PC with Windows Operating system, allowing the operator to conduct the following features

- 1.10.11. The operator shall be able to identify the category of an incoming call e.g internal, external, diverted, networked
- 1.10.12. The operator shall be able to extend calls automatically.
- 1.10.13. The operator shall be able to intrude in a conversation.
- 1.10.14. The operator shall be able to camp on a call on to a busy extension.
- 1.10.15. The operator shall be able to announce a monitored call.
- 1.10.16. The operator shall be able to transfer incoming calls to other operators.
- 1.10.17. The operator shall be able to mark an external call to be a serial call.
- 1.10.18. The operator shall be able to put the exchange into emergency state.
- 1.10.19. The operator shall be able to park calls and retrieve them later.
- 1.10.20. The operator shall be able to choose a specific line in a specific route.
- 1.10.21. Extended calls not answered within a certain time must be re-routed to the operator.
- 1.10.22. An extension shall be able to dial the operator by a general number and an individual number.
- 1.10.23. It shall be possible to group operators with respect to the traffic they are to
- 1.10.24. Telephony -System shall support IP Telephony for potentially future requirement based on:
 - 1.10.24.1. Remote Offices with IP Phones -It shall be possible to have Survivability for users of small Branch offices - without Needing a Local Gateway -System shall support Mobile extension as Backup for Remote IP Users so that Callers are not affected when they call Remote IP Phones in the case of IP Network Outages, and the calls made to such Extensions and DID Numbers shall mature without any problem using Integrated Mobile extensions..

- 1.10.24.2. Standard H.323 shall be supported for connection of IP phones and IP Soft clients.
- 1.10.24.3. Switching of a session between two IP-extensions shall be performed on the LAN /WAN to allow for different media like voice, video.
- 1.10.24.4. IP Telephones deployed in small branch offices (without local trunks) must support an alternative IP address for automatic failsafe operation to another IP Telephony gatekeeper in the event of WAN failure. This failsafe function must be automatic with no user intervention required.
- 1.10.24.5. Call Admission Control (CAC) shall be used to limit the number of concurrent calls per remote site, the number of calls being a figure set based on available bandwidth.
- 1.10.25. IP Phones support should be considered for future requirement- The Advanced IP Phones shall support the following minimum characteristics
 - 1.10.25.1. 2-port Ethernet switch (for sharing a LAN cable/port with a computer)
 - 1.10.25.2. User mobility (log-on and call)
 - 1.10.25.3. Registration with a default number function (the phone cannot be logged off)
 - 1.10.25.4. Emergency calls with location information
 - 1.10.25.5. Hands-free speaking feature with full duplex and Acoustic Echo Cancellation (AEC)
 - 1.10.25.6. Monitor speaker
 - 1.10.25.7. Headset port (with dedicated headset switching key)
 - 1.10.25.8. Power over LAN support (IEEE 802.3af compliant)
 - 1.10.25.9. User settings via web browser
 - 1.10.25.10. Hearing-impaired support
 - 1.10.25.11. Acoustic shock protection

- 1.10.25.12. Central storage of user settings for function keys
- 1.10.25.13. Call List (Local)
- 1.10.25.14. Multi-codec support G.711, G.723, G.729a G.729ab and GSM voice compression.
- 1.10.25.15. Multi-language support
- 1.10.25.16. Branch office Survivability support
- 1.10.25.17. Option Unit support (for connection of external betland/or busy signal)
- 1.10.25.18. The display shall be tilt able, with 6x50 characters.
- 1.10.25.19. 20 Function Keys.
- 1.10.25.20. Full Duplex Hands-free Speaker.
- 1.10.25.21. 3 Navigation Keys.
- 1.10.25.22. Up to. 4 extra key panels with 17 additional function key.
- 1.10.25.23. Phonebook must be stored locally on the phone or centrally on a server).
- 1.10.25.24. WAP browser with easy access to WAP portals, enabling fast access to WML Internet/intranet pages.
- 1.10.25.25. Menu support for absence handling (activation of diversion), personal number profile and absence reason message (message diversion).
- 1.10.25.26. The IP Phone shall be upgradeable by software downloading from a central HTTP server.
- 1.10.25.27. The phone must support full duplex hands-free operation and provide a separate headset port 1.9.25.28The IP Phone shall be self-configuring concerning IP-addresses and other pertinent data.

- 1.10.25.28. The IP Phone shall support the option of being centrally powered via the ethernet cable based on the IEEE802.3af version 3 standards.
 - 1.10.25.29. The IP phone should incorporate programmable keys, an on-board telephone directory (100 entries) and the ability to log incoming, outgoing and missed calls (50 calls).
 - 1.10.25.30. The IP Phone must support Quality of Service packet prioritization for speech and signaling according to Diffserv. Layer 2 quality of service according to IEEE802.1q, with VLAN tagging according to IEEE 802.1p & q.
 - 1.10.25.31. It must be possible to call emergency numbers i.e. 112, 911, without logging, on the IP Phone.
- 1.10.26. The IP Phones should support the IP monitoring/Boss Secretary feature allowing a person to pick a call for another person by just pressing a Function key programmed as MNS key, the boss may support multiple kind of extensions defined in his profile- IP/Analog/Digital/Mobile by pressing a single MNS key the secretary shall be able to take calls for the Boss.

1.10.27. IP soft-clients

The bidder should consider system that may support IP Client software to function as an IP "softphone." compatible with Windows 2000 / Windows XP.

The softphone must support all major SYSTEM features, including-

- Make Call
- Answer Call
- Hang up Call
- Call Back
- Last Number Redial
- Transfer Call
- Park/Retrieve Call
- Message Waiting Indication
- DTMF signaling

- Multiple Calls
- Call Logging (incoming, outgoing, missed)
- Call Status Display
- Volume control (including Mute/Un-mute)
- Secured Log-on

1.10.28. The softphone must integrate into centralized directories to provide the user with both took-up services and click-to-dial services.

1.11. System Integration With The Existing Telephone Facilities

1.11.1. All integration, interface and interconnection of equipment and the system shall have full and complete operability and interoperability and that the full and complete functional capacity and each individual subsystem shall not degraded in the process.

1.11.2. All information technology based systems and all interfacing, integration and connection systems shall have complete protection from software error.

1.11.3. Configuration for the integration with the END-USER existing telephony system and Internet connection shall be borne by the Contractor.

1.12. System administration, management and reporting

The system Administrations shall be conducted via a GUI based interface on a PC with capabilities to gather System Alarms and provide following information-

1.12.1 It should be possible to collect trunk and traffic data and generate reports.

1.12.2 There shall be different privileges for administration, maintenance and reports

1.12.3 All the changes and the results (success, failures) shall be logged with time indication.

1.12.4 It shall also be possible to prepare batch files which can be automatically sent to the SYSTEM at predefined times.

1.13. Diagnostics

1.13.1 The SYSTEM shall provide automatic self-diagnostic testing.

1.13.2 Diagnostic programs shall run continuously, 24 hours a day.

1.13.3 The alarms shall initiate a phone call to a service center.

1.13.4 The SYSTEM shall support remote diagnostics and software changes using an internal modem without the need for a dedicated line.

1.14. Reports

1.14.1 All the following standard system reports shall be available on request by the system administrator

- system performance
- network traffic
- call detail recording and relay for integration to a Billing system
- parameter settings
- configuration details

1.14.2 The system shall summarize usage for:

- a specific time period
- total number of calls
- busy hour traffic
- average call length

1.14.3 Call Information Logging shall be stored on a hard-disc for efficient retrieval; the same shall be processed on a system to provide call billing information.

1.15. Statistics

1.15.1 Individual port statistics shall be extracted from and displayed/printed by the system on demand.

1.15.2 Detailed user call statistics shall be extracted from and displayed/printed by the system in demand.

- 1.15.3 Detailed user facility statistics shall be extracted and displayed/printed by the system on demand.
- 1.15.4 The system's hardware and software error logs, indicating occurrences of system errors, shall be retrieved on demand.
- 1.15.5 All user and system statistics shall be further customized through a custom database report generator.
- 1.15.6 All reports and system statistic shall be retrievable while the system is processing.