# The Lawrence School

**Lovedale, The Nilgiris, Tamil Nadu - 643003**

# Tender Document

# for

# Intercom Telephony System

# Specifications for Telephony System

## 1 INTRODUCTION

This Specification brings out the qualitative and quantitative requirements of "IP PABX System".

The objective of this RFP in terms of architecture is unification of the voice, data and video communication infrastructures, to build-up a communication platform that is standards based.

The proposed solution would be based on IP Hybrid technology supporting IP/SIP, Digital and Analog end points.

Proposed solution infrastructure should deliver
- Highly Secured, Scalable and high reliable communications
- feature rich telephony over all end points

## 2 Communication System

a) The Contractor shall provide IP EPABX System, mountable in a standard 19" Network rack, that shall be non-blocking and open standard (based on H.323/ SIP), expandable up to 1200 ports on the same hardware. The solution shall support in-skin VoIP DSP channels at least 250 built-in from day 1 for non-blocking architecture per call server and should not consume any additional slot.

b) The proposed communication server should provide support for:
   - Media Gateways to allow TDM connectivity - **TRUNKS**
   - **ISDN PRI and BRI Interface**
   - **Analog Trunks Interface with CLI**
   - **E&M Trunk Interface (Type I – V)**
   - Media Gateways to allow TDM connectivity - **EXTENSIONS**
   - **Analog Extensions with CLI**
   - **Analog Long Line Extensions (Name & Extensions No.)**
   - **Digital Extensions with CLI (Name & Extensions No.)**
   - IP conversion and resources like conferencing, voice prompts.
   - Resources for IP Extensions & IP Trunks.
   - Local or remote network management systems

c) To ensure, complete ownership and smooth delivery of all the functional and reliability requirements of the solution proposed, all the hardware and software must be from the same OEM. No third party products or integrated solutions are allowed.

d) The proposed system shall be equipped to handle the required capacity but should be scalable to minimum 1,200 ports with no limitation to type of Analog / Digital / IP extensions or trunks from day 1.

e) System should evolve in the future into supporting multiple sites (Minimum 40), to ensure distributed / centralized architecture in a multi site environment. The system should support remote site architectures with self-survivable gateways and IP phones.

f) The system must be able to interoperate with other telephone systems and endpoints using the following standards:
   - SIP/ H.323
   - CTI
   - E&M

g) Proposed Communication server should be capable of handling peak hour traffic of up to 5000 Busy Hour Call Completion (BHCC) per server.

h) The system must support gateways for following external telephony interfaces and signaling protocols:
   - ISDN PRI
   - BRI
   - 2 wire and 4 wire E&M
   - Analog Loop Start and Ground Start (with CLIP)

i) SIP Carrier Profile – The offerred system must support minimum 6 SIP carrier profiles without any extra hardware.

j) Paging – The syste should built-in dual paging port to interface with the exisitng PA system for paging during emergency situations.

k) The system should support basic Unified Communication features i.e. Presence, BLF, IM and basic calling features without any external server for minimum 120 UC clients.

### 2.1.1 <u>Communication Server</u>

a) The proposed solution must be highly resilient and support avaialblity for the Communication Server.

b) Proposed architecture should have Primary – Secondary call server. Secondary servers shall act as independent unit in case network disconnection with primary server of IP PBX system.

c) The communication servers must always be in sync with the complete "mirroring" of fixed or variable data.

d) The IP System should have embedded feature of DHCP server to ensure reliable IP telephony functioning.

e) Media gaetways should also provide distributed power supplies to ensure reliable gateway architecture.

f) Gateway where all the extension and trunk interface boards are located should have enough switch matrix capacity to support high traffic. It should be equipped with minimum 90% timeslots avaialble for the number of ports equipped.

**2.1.2 <u>Integrated Media Gateways:</u>**

a) The system should support high density media gateway to support the TDM extensions and trunks.

b) It should support G723.1, G711, G722, G726, G729a Codecs.

c) The offered System should have minimum 250 VoIP DSP resources, universal slots, internal voice prompts with 16 simultaneous announcement, storage memory to accommodate 16 minutes of static prompts and 8 minutes of dynamic prompts, 2 groups of 30 party meet me conference.

**2.1.3 <u>System features</u>**

a) The IP PBX should support flexible numbering plan with upto 8 digits. System should allow mixed numbering scheme.

b) The system should provide distinctive ringing for internal calls, junction calls, auto call back, wake up service and emergency reporting service.

c) The proposed system should support automatic route selection (ARS) and least cost routing (LCR) features to route the calls based on priorities related to user profile, tariff, and network availability, along the most cost-effective path. This service will be transparent for users and irrespective of the physical carrier connection. It should be possible to inform the caller via voice prompt if the cheapest route is not available.

d) The proposed system should have both live and automated attendant's features and either of the one or combination of both can be deployed to fulfill the requirement. At least 16 ports should be provided in the system.

e) The system should support voice prompts to guide the user in all communication positions by multilingual voice guides. Minimum 64 context sensitive Voice prompts should be part of each system.

f) The system should have the following cost control features.
- Class of service
- Forced account code
- Authorization code
- Least cost routing
- Toll restriction
- Automatic route selection
- Call duration timer/warning

g) It should be possible for a user to make an external call from other sets via a PIN thus bypassing the external call barring category for that set.

h) Hunt groups – Sequential, Cyclical or "Parallel". The telephone sets in the group could be analog phones, IP phones or a combination of both. In case of overall occupancy, the calls will stay on hold in a waiting queue, or be redirected towards another user, station group, attendant or an external correspondent. Total no. of group should be 64. It should be possible to have up to 64 extensions in one group.

i) **The system should support the following basic telephone features**:

**i.** Abbreviated dialing
**ii.** Appointment reminder
**iii.** Automatic call-back on busy trunk/bundle/network link
**iv.** Call forwarding unconditional on busy/no reply to extension, hunting group, Voice mail, operator, paging, etc.
**v.** Conditional external forwarding (busy or no reply)
**v.** External call forward to PSTN/Cellular .
**vi.** Immediate forwarding
**vii.** Call pick-up
**viii.** Call parking
**ix.** Call tracing
**x.** Call waiting indication / voice prompt
**xi.** Calling line identification restriction for internal calls
**xii.** Calling party identification for internal users.(on analog phones also).
**xiii.** Camp on busy telephone/hunting group/voice mail
**xiv.** Controlled private call by Pin code and password
**xv.** Do not disturb
**xvi.** Dynamic call barring
**xvii.** General night service
**xviii.** Hunting group (fix head, cyclic, longest idle time, parallel)
**xix.** Internal/external music on hold
**xx.** Internal/external inquiry call
**xxi.** Individual hold
**xxii.** Instrument locking to prevent the outgoing calls.
**xxiii.** Last internal/external number redial

xxiv. Store and redial external number

xxv. The system must support 32 party x 2 groups conference from day one.

xxvi. The system must have minimum 96 conference resources.

xxvii. The system should support multiple conference options. i.e. Adhoc, Meet-me, Dial-in conference with Password Protection.

xxviii. Multiple appointment reminders per extensions

**j) The system should also support the following telephone features on Digital sets in addition to above basic features:**

- Adjustable handset volume
- Call-by-name for internal and external user.
- Centralized phone book stored in communication server directory feature with spelling and syntax corrector with 1000 entries, accessible from the alphabetical keyboard of the phone. Caller name display

- Conditional/unconditional differentiated forwarding of multiple directory numbers
- Contextual voice prompts
- Distinctive ringing.
- Fixed function keys
- Hands-free
- Headset capability
- Individual customization
- Interactive guidance with soft keys
- Loudspeaker announcement
- Multi-line: multi-appearance and multi-numbers
- Multi-line selective forwarding
- Message waiting indication
- Text messaging for mid range and high end digital and IP phones
- Calling line identification presentation/restriction (CLIP/CLIR)
- Digit by digit dialing mode
- End block dialing (digit correction possible)
- ISDN identification (CLIP) converted into name
- ISDN mini-text messages (carrier dependent)
- Malicious call identification
- Storage of unanswered calls with date, time, and callback

### 2.1.4 Network Services

a) As this project involves multiple locations up to 49 sites deployed with independent system and networked over PRI / IP, the network should support complete feature transparency to get the maximum benefit from the latest technologies

b) Same set of protocols should be used for PRI / IP networking of the systems and they should provide transparent services across the network.
c) List of network wide features to be systematically supported across the network over PRI / IP are list below:

- Basic call
- Number and name identification
- Call/dial by name
- Call waiting indication
- Call offer
- Callback on busy links
- Callback on free or busy extensions
- Call park
- Call forwarding
- Camp on
- Individual call pick-up
- Data call
- Distinctive ringing
- Hold
- Intercom call
- Consultation call
- Broker call
- Transfer
- Conference
- Intrusion/Barge-in
- Unconditional forwarding
- Conditional forwarding
- Do not disturb
- DISA
- Transparency of rotary and DTMF dialing
- Retransmission of last number dialed (redial)
- ISDN/PSTN supplementary services
- Multi Party Conference
- Announcement/paging on speaker
- Manager/assistant filtering/screening team
- Object supervision: free, partially busy, totally busy, ringing

- Hunt groups
- Centralised Attendants and Voice messaging serving the entire network

### 2.1.5 Voice Messaging application

a) Voice messaging application has to be from the same OEM as that of the IP PBX and should support unified messaging features.

b) The offered Voice messaging application should be a single system delivering both standard voice messaging as well as unified messaging services.

c) Voice Mail features

- Messaging facilities
    - Direct access to the mail box (by-pass the menu greeting) from business phone set (IP or analog) and from any phone set if the phone number is declared by the user in his personal settings.
    - Internal call routing (busy and no answer)
    - Maintain a specific number of minutes of messages per user
    - Guest mailboxes
    - Mailbox escape to predefined extension
    - Future delivery
    - Automated attendant
    - The messaging application must have the ability to record live conversation through press of a button on the IP/digital phone or through the softphone application.
    - Web based interface for message consultation, play, deletion

- Notification facilities
    - Message notification on phones, emails
    - Message waiting audible voice prompts
    - Integration with visual message waiting indicators on phones
- Personal settings facilities
    - Password management
    - Call forwarding direct to mailbox
    - Password protected mailbox
    - Personalized voice greetings (busy, no answer)
    - Private messages
    - Announcement - only mailbox
    - Visual call management on display

- Message Listening facilities
    - Playback control
    - Message header information
    - Message-rewind, pause and fast-forward
    - Reply to sender
    - Send a message copy
    - Skip messages

### 2.1.6 Attendant (Operator) positions

a) Proposed solution should be Phone Based Operator console. Phone based operator should have minimum 24 keys on the set and should support Add on modules with atleast 60 keys from day.

b) Attendant Console should support following :

- Override **-** The attendant can bypass forwarding or do not disturb modes on a phone
- Called Party Call Resources **-** When the called party is busy or does not answer, the following call resources should be available on attendant keys:
    - Callback request
    - Text message
    - Voice mail

- The attendant should be able to send mini text messages to a users with display:
    - Programmed message
    - Pre-programmed message completed by entering digit: date, hour
    - Pre-programmed message completed by any variable part (text and digits)
    - Fully programmable message (minimum twenty characters)

- Transfer on no Answer
- Transfer on Busy
- Barge-in on Busy or Partially Busy
- Three-party Conference
    - An attendant must be able to set up a three-party conference, and then transfer the conference to another user.
- Call Transfer to other Attendant
- Serial Call
- Recall
- Class of Service **-** The attendant console (in idle state) must be able to change the class of service for any phone.

- Public Trunk Access Restriction - The attendant console (in idle state) must be able to change the public trunk access and restriction for any phone.
- Directory access and call by name

### 2.1.7 <u>Digital Telephone terminals</u>

All phones should be from the same OEM as the PABX.

**a) Type 1: Entry level Terminals**

- 3-line 20-character B&W display
- Hands-free and amplified listening modes with volume control
- Dual line
- Six programmable keys
- Nine fixed feature keys
- Directory Dial- 1000 System Numbers, 10 Personal, 500 Phone Book
- Mute
- Redial
- Message lamp with Multi Colour LED

**b) Type 2: Mid range Digital Phones**

- Adjustable B&W graphical display with minimum LCD size of 168 x 58 dot matrix Tiltable display.
- 4 physical soft keys associated with LCD with key content display, should support atleast 10 fixed feature keys.
- 4 way Navigation key with exit and validation keys for use with the graphical interface
- 12 one touch programmable keys with LEDs
- Full Duplex Hands free and group listening modes
- Message wait lamp with Tricolour LED
- RJ9 connection
- Should support Add on modules to add minimum 60 keys
- Terminal should support Analog port adaptor for ringer and modem functionality and Recording Adaptor for conversation recording.

**c) Type 3: High end IP Phones**

- Should support following user interface features:
- 7 in TFT LCD display 1024 x 600 with backlight, 16.7M Colours.
- Capacitive 4 finger multi-touch window.

- Flexible through adjustable height.
- Good voice quality for each audio path-handset, the hands free speakerphone and blue-tooth headset
- Full-duplex conversations, acoustic echo cancellation and background noise suppression
- Frequency response – Wideband audio for handset, blue tooth headset and hands-free speakerphone modes
- Dedicated RJ-9 headset port and blue-tooth support
- POE enabled
- Micro SD Card support
- USB support
- Flexible line appearance (one or more line keys can be assigned for each line extension)
- Distinctive incoming call treatment/call waiting
- Call timer and call waiting
- Call transfer, hold, divert (forward), pickup
- Called, calling, connected party information
- Should be able to participate in audio conferencing as part of the overall solution
- One-touch speed dial, redial
- Video
- RFC3261, RFC3550, RFC3711, RFC4566- SDP
- Should support following Network and provisioning features:
- SIP Protocol
- Dual Ethernet Gigabit Port with Dual switched autosensing mode.
- 10/100/1000Base-TX across LAN and PC/Laptop ports
- Manual or dynamic host configuration protocol (DHCP) network setup
- Time and date synchronization using NTP/SNTP
- QoS Support – IEEE 802.1p/Q tagging (VLAN), Layer 3 TOS, and DSCP
- Status and statistics reporting through centralized management
- Should support following Security features:
- Media encryption
- Password login

## 2.2 ISDN based Video conference system interface

1. System should support with 32 numbers of SO ports for interfacing ISDN based video conference system.
2. The offered system should support in-skin Video MCU with minimum 4 party multi-conference.
3. The system should be compatible with in-skin web based UC conference (8 party per group) from day 1.

## 2.3 Security

a) Security is currently one of the most important needs when a VoIP project should be deployed inside an enterprise. **The Tenderer must only propose an offer that conforms with adequate security standards. The quoted system should be supported with SBC from same OEM with below features:**

- SIP Trunking
- Remote worker
- SIP registration passthorugh
- SIP intrusion prevension: 400+ SIP attack signature support
- DDoS / DoS attach protection
- SIP registration scan attack detection
- SIP header normalization
- SIP malformed packet protection
- Topology hiding with SIP deep packet engine
- Call security with TLS
- RTCP statistics
- T.38 Fax relay
- SIP Firewall – Should support deep packet inspection protection at Layer 7.
- VLAN and virtual IP support
- Minimum 30 concurrent channels with thorughput 300Mbps
- Local security event console
- Remote syslog
- Device management: Web GUI via Https & SSH CLI

**b) Communication server security**

i. Controlled access to management platform is imperative. The system must control the identity of the management terminals and the user accessing that terminal. During a connection, (local or remote) the system must check the consistency between the management platform name, management platform password, and user name before authorizing the connection.

ii. The System must support Network Time Protocol V3 or higher to synchronize the system data/time of network devices

iii. The call server Operating System must be hardned Linux and it must not use or natively support network resource sharing services such as NFS, Samba or LPR.

iv. To avoid the introduction of virus, worm and Trojan type attacks; the use of internal e-mail servers is not acceptable.

v. The Call Server must not employ the use of a 'default' password that is viable beyond the period of installation.

vi. The password & access control must include at least:
- Shadow Passwords to prevent the possibility of an aggressor to easily read or deduce system or account access passwords.
- Usage of MD5 algorithm (or stronger) for password encryption

vii. Denial of Service
- Proposed EPABX and Media gateways must provide self-protection mechanisms to counter this type of attack.
- Media Gateways should not host services such as proxy, FTP, Telnet or local dynamic routing to prevent exploitation in Distributed Denial of Service attacks.

**c) Management Security**

i. Role Based Account Management to define different levels of administrator access depending on specific function responsibility.

ii. Administration users connecting directly to the Call Server (console) and management platform must be authenticated before gaining access to the call server.

### 2.3.1 <u>Call accounting</u>

i. CDRs should be generated for internal and external calls for incoming and outgoing calls with the following fields:
- Caller / called termainal name/ number
- Number dialed (with masking of the last four digits) / caller number
- Date, hour, and minute that the call ended
- Length of the call
- Number of cost units
- Cost for the call
- Type of call
- Log for walking class of service.
- Account code

### 2.4 Environmental Conditions

The equipment offered shall be capable of maintaining its guaranteed performance when operating continuously for 24 hours a day and 365 days a year under the following environmental conditions.

- Operational temperature : 0 to 40 Degree C.
- Humidity : 10% to 90% RH (non-condensing)

## 3 Bill of Materials :

| Sl No. | Description | Qty |
|---|---|---|
| A | The solution shall be a hybrid PBX with both Time Division Multiplexing (TDM) and IP telephony with trunk support, used in either mode or both concurrently The contractor shall provide NEC IP EPABX System, mountable in a standard 19" Network rack, that shall be non-blocking and open standard (based on H.323 and SIP), expandable to 1200 ports on the same hardware. The solution shall support in-skin VoIP DSP channels at least 250 built-in from day 1 for non-blocking architecture per call server and should not consume any additional slot. The system shall be equipped with the following: | NEC IP SERVER SV-9100 |
| 1 | Voice Call Server | 05 |
| 2 | ISDN PRI Interface (30 Channel) | 01 |
| 3 | Analog Trunk Circuits with integrated FSK & DTMF CLI support | 08 |
| 4 | Netlink LICENSE | 07 |
| 5 | Digital Extensions Circuits | 08 |
| 6 | IP Extension License | 08 |
| 7 | Analog Extension Circuits with Extensions & Name display | 304 |
| 8 | DSS 60 One touch access keys for trunks & extensions | 02 |
| 09 | 16 Port Voice response system with 40 hrs built-in storage of voice Mail | 01 |
| 10 | 300 Pair IPM | 1 SET |
| 11 | 19” 42U Server Rack (600x800) with 4 Fans, Castor, PDU, Cable Manager, Glass door and Hardware | 01 |
| 12 | NEC Digital phone 24 key | 01 |
| 13 | NEC IP Phone 24 key | 08 |
| 14 | Installation, Commissioning & Programming Charges | 01 |
| 15 | Product Warranty Details | - |
| 16 | Buy back @ Old Epabx “NEC” 8100 (220 Extension with 5 Servers) | - |