वर्गीय आवश्यकताओं के लिए मानक

टीईसी 60030:2016

(पूर्व सं: टीईसी/जीआर/एसडबल्यू/पीबीएक्स-005/01/सितंबर16)

STANDARD FOR GENERIC REQUIREMENTS

TEC 60030:2016

(Earlier No: TEC/GR/SW/PBX-005/01/SEP-16)

मीडिया गेटवे के साथ आईपी पी.ए.बी.एक्स.

IP PABX with MEDIA GATWAY

ISO 9001:2015

दूरसंचार अभियांत्रिकीकेंद्र

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FOREWORD

Telecommunication Engineering Centre (TEC) is the technical arm of Department of Telecommunications (DOT), Government of India. Its activities include:

- Framing of TEC Standards for Generic Requirements for a Product/Equipment, Standards for Interface Requirements for a Product/Equipment, Standards for Service Requirements & Standard document of TEC for Telecom Products and Services
- Formulation of Essential Requirements (ERs) under Mandatory Testing and Certification of Telecom Equipment (MTCTE)
- Field evaluation of Telecom Products and Systems
- Designation of Conformity Assessment Bodies (CABs)/Testing facilities
- Testing & Certification of Telecom products
- Adoption of Standards
- Support to DoT on technical/technology issues

For the purpose of testing, four Regional Telecom Engineering Centers (RTECs) have been established which are located at New Delhi, Bangalore, Mumbai, and Kolkata.

ABSTRACT

This document specifies the Generic Requirements (GR) of 'IP-PABX with Media Gateway' connected to Indian Telecom Network. This document covers the Generic Requirements of IP-PABX for supporting speech communication, data communication and multi-media applications. It covers its facilities, features and performance requirements.

CONTENTS

HISTORY SHEET

Sl. No.	Standared GR No.	Title	Remarks
1.	TEC 60030:2016 (Earlier No: TEC/GR/SW/ PBX-005/01/SEP-16)	Generic Requirement of IP-PABX with Media Gateway	Issue 01
Document number changed as per Revised Numbering scheme of TEC for conversion of existing TEC document to Standard vide document no.4-47/2019-RC/TEC dated 07-09-2019			

Note:

1. Since the documents have been renumbered as per revised numbering scheme, kindly refer the Mapping- Listing Table pertaining to old and revised document number available on TEC website www.tec.gov.in/. In case of further clarification, please contact at e mail id adgdoc.tec@gov.in
2. Inside the document, GR may be read as Standard for GR, IR as Standard for IR, SR as Standard for SR and TSTP as TEC Test Guide."

REFERENCES

S. No.	Document No.	Title/Document Name
(I)	TEC GR/IRs	
1.	TEC/EMI/TEL-001/01/FEB-09	EMI/EMC Standards
(II)	ITU Standard/Recommendations	
1.	ITU-T G.114	One-way transmission time
2.	ITU-T G.168	Digital network echo cancellers
3.	ITU-T G.703	Physical/electrical characteristics of hierarchical digital interfaces
4.	ITU-T G.711	Pulse code modulation (PCM) of voice frequencies
5.	ITU-T G.726	Coding of analogue signals
6.	ITU-T G.729	Coding of voice and audio signals
7.	ITU-T T.38	Procedures for real-time Group 3 facsimile communication over IP networks
8.	ITU-T H.264	Advanced video coding for generic audiovisual services
9.	ITU-T Q.68	Technical features of push-button telephone sets
10.	ITU-T Q.920	ISDN user-network interface data link layer – General aspects
11.	ITU-T Q.921	Data link layer specification
12.	ITU-T Q.931	ISDN user-network interface layer 3 specification for basic call control
(III)	IETF Recommendations	
1.	IETF RFC 768	User Datagram Protocol (UDP)
2.	IETF RFC 791	IPv4 addressing
3.	IETF RFC 793	Transmission Control Protocol (TCP)
4.	IETF RFC 1034	Domain names – Concepts and Facilities
5.	IETF RFC 1305	Network Time Protocol
6.	IETFRFC 2119	Key words for use in RFCs
7.	IETF RFC 2326	Real Time Streaming Protocol (RTSP)
8.	IETF RFC 2327	SDP: Session Description Protocol
9.	IETF RFC 2460	IPv6 addressing
10.	IETF RFC 2782	A DNS RR for specifying the location of services (DNS SRV)

11.	IETF RFC 2806	URL'S (Uniform Resource Locator)for Telephone calls
12.	IETF RFC 2915	The Naming Authority Pointer (NAPTR) DNS Resource Record
13.	IETF RFC 2916	E.164 number and DNS
14.	IETF RFC 3261	SIP: Session Initiation Protocol
15.	IETF RFC 3262	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
16.	IETF RFC 3263	Session Initiation Protocol (SIP): Locating SIP Servers
17.	IETF RFC 3265	Session initiation Protocol (SIP) – Specific Event Notification
18.	IETF RFC 3311	The Session Initiation Protocol (SIP) UPDATE Method
19.	IETF RFC 3389	Real-Time Transport Protocol (RTP) Payload for Comfort Noise (CN)
20.	IETF RFC 3550 & 3551	Real Time Transport Protocol (RTP, RTCP)
21.	IETF RFC 3711	SRTP (Secure Real-time Transport Protocol)
22.	IETF RFC 3761	E.164 Uniform Resource Identifier(URI) Dynamic Delegation Discovery System (DDDS) Application (ENUM)
23.	IETF RFC 4733	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
(IV)	Other Standards	
1.	CISPR 11	Limits and methods of measurement of radio disturbance characteristics of industrial, scientific & medical (ISM) radiofrequency equipment
2.	CISPR 22	Limits and methods of measurement of radio disturbance characteristics of ITE
3.	EN 55011	Industrial, scientific and medical (ISM) radio-frequency equipment - Electromagnetic disturbance characteristics - Limits and methods of measurement
4.	EN 55022	Information Technology Equipment - Radio disturbance characteristics - Limits and methods of measurement
5.	IEC/EN 61000-4-2	Testing and measurement techniques – Electrostatic discharge immunity test
6.	IEC/EN 61000-4-3	Testing and measurement techniques – Radiated, radio-frequency, electromagnetic field immunity test
7.	IEC/EN 61000-4-4	Electromagnetic compatibility (EMC) - Part 4-4: Testing and measurement techniques - Electrical fast transient/burst immunity test
8.	IEC/EN 61000-4-5	Electromagnetic compatibility (EMC) - Part 4-5: Testing and measurement techniques - Surge immunity test

9.	IEC/EN 61000-4-6	Electromagnetic compatibility (EMC) - Part 4-6: Testing and measurement techniques - Immunity to conducted disturbances, induced by radio-frequency fields
10.	IEC/EN 61000-4-11	Electromagnetic compatibility (EMC) Part 4-11: Testing and measurement techniques Voltage dips, short interruptions and voltage variations immunity tests
11.	IS 10437 / IEC 60215	Safety requirements for radio transmitting equipment
12.	IS 13252 part 1:2010 / IEC 60950-1 {2005}	Information Technology Equipment -- Safety, Part 1: General Requirements

CHAPTER-1

1.0 Introduction

1.1 Private Automatic Branch exchange (PABX) is an in-house telephone switching system which makes connections among the internal telephones of a private organization and also connects them to the public telecom network via various interface(s) as depicted in Fig- A.1 of this document.

1.2 In 'IP PABX' call processing functions are performed by the server and IP devices (e.g. SIP phone, PC) and Ethernet interface to IP network is provided directly through LAN switch. To provide TDM type extension (e.g. analogue phone, digital Phone) and interface to TDM network, media conversion functions are performed by Media gateway. This document covers the Generic Requirements of 'IP-PABX with Media gateway' for supporting speech communication, data communication and multi-media applications. It covers all types of interfaces for connection to public network as well as essential IP-PABX facilities, features and its performance requirements.

1.3 Approval of equipment against this GR, shall not be construed as an authorization to evade surreptitiously, regulations including toll-bypass concerning the use of telecom services. Functioning or intended use of the equipment shall conform to the prevailing laws/regulation/instructions of Govt. of India.

1.4 Approval of IP-PABX against this GR, does not entitle the user to connect the equipment to the network of Internet Service Provider's (ISP).

1.4 (a) For all ITU–T/IEEE recommendations, TEC standards/specification and other standards referred in this document, the latest release/issue with all associated amendments, addendum and corrigendum shall be applicable.

(b) The RFC's documents of IETF are subject to periodic revision. Hence, where ever RFC's are mentioned in this document, the offered product shall meet either the referred RFC or its previous version or its previous draft or its updated version. Wherever a feature of RFC is mentioned, the product shall comply with the part of RFC specifying the feature.

(c) For all IETF RFC's, the interpretation of clauses of RFC's shall be as per RFC 2119

1.5 All the requirements described in chapter 2 of this document are suggestive requirements and shall be decided by the purchaser at the time of procurement/tender as per its requirements. However, the requirements described in Chapter-2 will not be tested/ verified by TEC at the time of initial certification.

2.0 Description

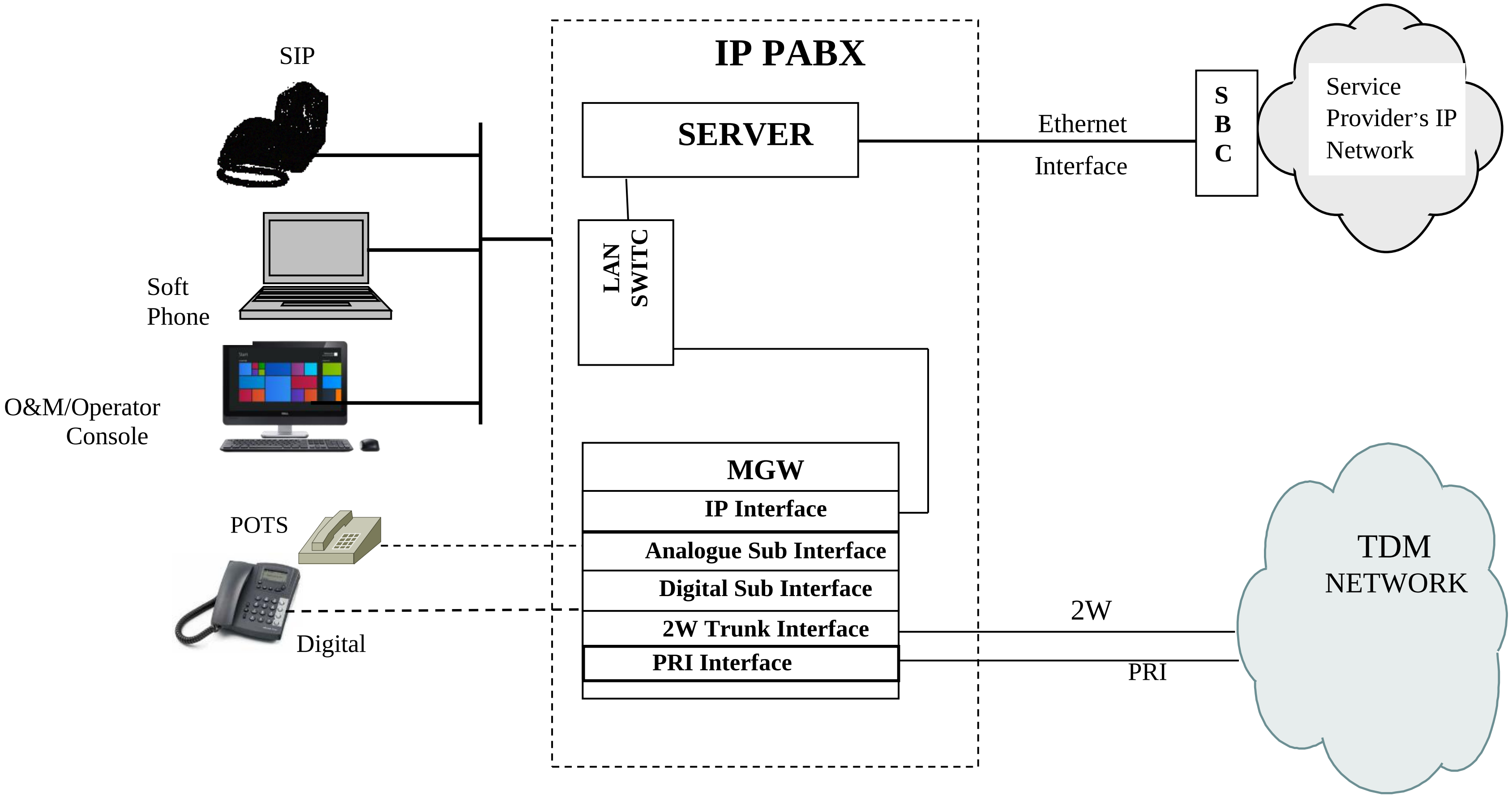
2.1 'IP PABX with media gateway' can be a single box (COTS or an Embedded Server Card) or multiple box entities. The system may consist of different types of entities such as SIP server/Call Server/Call manager, Media Gateway, MGC, Signalling gateway etc. MGC, Signalling gateway functionalities can be in built in SIP server/Call Server/Call manager or Media Gateway or can be separate entities. LAN switch may or may not be part of IP-PABX systems. This GR does not specify the LAN switch/MGC/Signalling gateway capabilities. (A typical block diagram of 'IP-PABX with media gateway' system is shown in Figure-A.1)

2.2 It allows IP user of an enterprise to connect to another user of same IP-PABX as well to external phone lines. It can also switch calls between an IP user and a traditional telephone user. This means VoIP communications and traditional telephone communications are all possible using a single system.

Figure-A.1

(Simple Block Diagram)

IP-PABX with media gateway



3.0 Functional/Operational Requirements

3.1 This GR is intended to facilitate the verification of the capability of the IP-PABX with Media Gateway for correct inter working with Indian Telecom networks.

Type I interface i.e. Ethernet interface which shall be connected to IP core network is mandatory. Additionally, the applicant may seek type approval for Type-II or Type-III or both types of interfaces given below:-

i.	Interface TYPE I	Ethernet interface
ii.	Interface TYPE II	2W DEL: 2-wire DELs for outgoing calls as well incoming calls. The junctions of IP-PABX will be terminated on the analogue subscriber line cards of the main exchange
iii.	Interface TYPE III	ISDN Primary rate access interface (PRI i.e. 30 B + D) as per TEC standard No. SD/ISN-01 and shall terminate on the ISDN PRI access card.

3.2 All the functional requirements described in clauses 3.2, 3.3 & 3.4 and their sub-clauses, shall be complied by all IP-PABX irrespective of types of interfaces and/or the type of extensions for which approval is sought. Additional requirements for different types of interfaces/extensions are given in subsequent clauses of this document.

3.2.1 The applicant shall submit the list of all units which are part of IP-PABX for which the type approval is sought. The equipment shall be complete in itself along with associated power supply arrangement.

3.2.2 The calls within IP-PABX system shall be switched without using link with Indian Telecom Network. Intra-PABX calls shall not be affected in case of link failure to Indian Telecom Network. When the links between Public network and IP-PABX are out of service, IP-PABX shall send suitable announcement/tone/message to calling extension subscriber.

3.2.3 It shall be possible to make calls to emergency numbers (e.g. 100, 101, 102 etc.) and IN calls from all types of user terminals i.e. extensions provided. For O/G calls, in the post dial scenario, IP-PABX shall support transmission of DTMF as per ITU-T Recommendation Q.23 in case of TDM interface (i.e. type-II & type-III) and as per IEEE Recommendation SIP info or RFC4733 in case of Ethernet interface (i.e. type-I)

3.2.4 The system shall support SIP extensions (e.g. SIP phone) and Ethernet Interface. Additionally, it may support analogue/Digital extensions.

3.2.5 The system shall allow direct registration/profile creation of IP endpoints onto it and perform all functions of Proxy/ Registrar/Redirect etc.

3.2.6 The system architecture shall allow for incremental application without modification to existing feature.

3.2.7 Call processing – Call processing function shall take place in the server. Intra IP-PABX calls between IP terminals and calls from IP extension to public network via Ethernet interface shall not be affected in case the media gateway is powered off/faulty.

3.2.8 Addressing: It shall support IPv4 as well as IPv6 extensions (one at time). It shall have the capability to inter-work with IP networks supporting IPv6. Optionally it may interwork with network supporting IPv4.

3.2.9 The IP-PABX shall support calling the PSTN telephones by their E.164 address. For Intra PABX calls from IP terminals, it shall be possible to call IP terminals by IP address also.

3.2.10 The IP-PABX shall provide for the facility of the extension user to transfer the incoming call from the main exchange to other extensions directly (i.e. without the assistance of IP-PABX attendant/operator).

3.2.11 The IP-PABX shall have capacity to send and/or receive DTMF tones in RTP packets and relay these signals to physical terminations. (For IP to analogue & analogue to IP calls only)

3.2.12 IP-PABX shall have inbuilt all tones/announcement required.

3.3 Management Functionalities:

3.3.1 The IP-PABX system shall be manageable through a management interface. This interface may be built in the system or may be a standalone application. IP-PABX shall provide Graphical User Interface (GUI), for system management & maintenance.

3.3.2 O&M terminal shall support Graphical User Interface (GUI) for maintenance, configuration, management and supervision. It shall be possible to test user's extension/trunk circuits/IP link from O&M terminal.This O&M terminal functionality can be incorporated in system or PC/Laptop or standalone terminal.

3.3.3 It shall be possible to configure the IP-PABX from remote Web based interface. It may be done either by using O&M terminal or by remotely connected PC/Laptop.

3.3.4 The fault log can be stored in the IP-PABX system or can be sent to any of the network computer for backup.

3.3.5 The management platform shall support following management functions:

3.3.5.1 Configuration Management - The management system must be able to configure various services, users, class of service for all users, all system parameters and features.It shall be possible to change the configuration of the system and do troubleshooting.

3.3.5.2 Terminating Free Number: It shall not be possible to make call from public network to any extension as charge free call.

3.3.5.3 (a) Subscriber Data Record- IP-PABX shall provide Subscriber Data Record (SDR) for all calls i.e. Intra-PABX calls as well as calls to/from Public network. Facility shall exist to take a print out for each individual extension.

(b) At least the following information shall be given in a SDR:

i Type of call (originating or terminating)

ii Date and time of answer (in case of successful call)

iii Calling subscriber identity.

iv Socket Address i.e. Port no. & IP address of calling subscriber. (for calls made from IP terminals only)

v Incoming/Outgoing junction number

vi Number dialled by the calling subscriber i.e. called number

vii Duration of conversation in multiple of seconds.

viii Indication of Type of supplementary services used during the call (e.g. normal call, conference call, call forwarding etc.).

ix Redirecting number in case of call forwarding.

(c) Provision shall exist for storing the SDR data on the system and transfer the same on any other external device/system. Applicant shall declare the SDR storing capacity of the system.

(d) Applicant shall supply the required information in the human readable format and the 'data structure' to monitoring agencies. Required software may be part of the IP-PABX system or may be supplied as an external entity. Neither it shall be possible to edit the SDR file nor, it shall be possible to delete the file from the system.

3.3.5.4 Interface to Directory Services: It shall be possible to connect IP-PABX system to any LDAP protocol based server to get the directory information. IP-PABX shall use any standard directory protocol (LDAP etc.) or propriety protocol to send the directory to its IP phone users.

3.3.5.5 Faults and Alarm–

a. The offered Management platform shall have provision for storing all alarm and event messages date wise in a separate file and shall have the capacity to analyze the same. It shall also be possible to access/route these faults and alarm messages on any of the operator console/ maintenance terminal.
b. It shall provide facility for isolating the faults on the IP-PABX side or exchange side for any particular trunk/ link and/or all trunks/ links.
c. In case any faulty condition of any trunk/link detected by the Management platform, a suitable alarm/indication for the particular trunk/link shall be provided.
d. It shall be possible to test the trunk/ link from main/parent exchange of service provider.

3.3.5.6 Performance Management –

(a) Processor Occupancy Report: The system shall offer processor occupancy report indicating the CPU usage, memory usage and uptime.
(b) Performance Reporting: The management platform shall be able to generate reports in terms of error reports and alarm reports. It shall be possible to export information on to an excel format.
(c) The management platform shall be able to interact with the Call control server for all backup related services. It shall be possible to keep back-up or archive of the system on any external device/system.
(d) The management system shall be compliant to SNMP v3 or the latest version.

3.3.6 Command log - When any command is executed, its response message shall contain the date and time. Commands which are used for modification of IP-PABX program or data shall be stored in the system. Provision shall exist for storing the command log/history log on the system or any other external device/system. It shall not be possible to modify or delete log file. Applicant shall declare the storing capacity of log file in the system.

3.3.7 System back-up- It shall be possible to save system back-up/history log in the system storage. In addition, it shall also be possible to save system back-up/ system log in external device/system. It shall also be possible to load the system from system back-up.

3.4 IP-PABX system shall support SIP terminals extensions. Applicant may support one or both type(s) of extensions given below:-

i. SIP Hard Phone

ii. SIP Soft Phone

3.4.1 SIP hard phone shall be connected to IP-PABX using RJ45 connector. (Applicable only if applicant has sought approval for SIP hard phone).

3.4.2 It shall be possible to load SIP phone software client on any PC. (Applicable only if the applicant has sought approval for SIP soft phone).

3.4.3 In addition to SIP phone, IP-PABX may support other type(s) of extensions such as analogue phone, digital phone etc.

3.5 Functional Requirements (Applicable for Type-I i.e. Ethernet Interface only)

In addition to functional requirements described in clauses 3.1 to 3.4 of this document, this clause and its sub-clauses specifies the functional requirements for IP-PABX with Ethernet interface using SIP signaling.

3.5.1 The IP-PABX shall have the capability to interconnect with any switching Node (Media Gateway or Soft-switch or CSCF).

3.5.2 It shall be possible to connect IP-PABX to IP network (Soft switch or Media Gateway or CSCF) through Session Border Controller (SBC), which shall be located at the administrative boundary of the IP network. SBC shall be deployed at the edge of core of a service provider’s network to control signalling and media streams as they enter and exit the network. Sub-systems of the SBC communicate to one another over IP interfaces.

3.5.3 Failures in the IP-PABX causing inability to receive incoming calls shall be signalled from the IP-PABX to the public exchange/ network by suitable means.

3.5.4 The table on below shows the complete Ethernet frame (this is as per the IEEE 802.3 standard) for the MTU of 1500 bytes

<table>
<tr><th colspan="9">802.3 MAC Frame</th></tr>
<tr><th>Preamble</th><th>Start-of-Frame-Delimiter</th><th>MAC destination</th><th>MAC source</th><th>802.1Q header (optional)</th><th>Ether type/Length</th><th>Payload (Data and padding)</th><th>CRC32</th><th>Inter frame gap</th></tr>
<tr><td>7 octets of 10101010</td><td>1 octet of 10101011</td><td>6 octets</td><td>6 octets</td><td>(4 octets)</td><td>2 octets</td><td>46–1500 octets</td><td>4 octets</td><td>12 octets</td></tr>
<tr><td colspan="2"></td><td colspan="6">64–1522 octets</td><td></td></tr>
<tr><td colspan="8">72–1530 octets</td><td></td></tr>
<tr><td colspan="9">84–1542 octets</td></tr>
</table>

3.5.5 After a frame has been sent, 12 octets of idle characters shall be transmitted before transmitting the next frame.

3.5.6 Calling Line Identification

(a) For all types of calls received from Public network, Calling Line Identification (CLI) received shall be transmitted to called extension. It shall not be possible to change CLI.

(b) For all types of calls made from extension to public network, CLI of extension number shall be sent to the main exchange.

3.6 Functional Requirements (Applicable for Type-II i.e. 2W DEL Interface only)

In addition to functional requirements described in clauses 3.1 to 3.4 of this document, this clause and its sub-clauses specifies the functional requirements for IP-PABX with 2W DEL interface.

3.6.1 The IP-PABX shall have the capability to interconnect with the Public Network Exchange (also referred as main exchange in this document) using 2-wire analog subscriber access interfaces (DELs) for originating as well as terminating calls, The junctions of IP-PABX (i.e. DELs) will be terminated on the analog subscriber line cards of the public exchange.

3.6.2 Whenever any extension goes off-hook, the IP-PABX shall feed dial tone to the extension.

3.6.3 Access to Public Network: The Public Network shall be accessed by dialing single digit. Dialling of `0' (or any other suitable code) shall be used to provide access to the Public network (O/G junctions) and the junction access should have a random or rotating priority to distribute the traffic. Normally, after receipt of digit ‘0’, IP-PABX shall put through the call to main exchange which shall feed second distinctive dial tone to calling extension subscriber. Further digits shall be dialled by the caller after receipt of second dial tone.

3.6.4 Line Hunting: For outgoing calls, the IP-PABX shall provide access to the public network using sequential or random search for selection of trunk/link towards Indian Telecom Network.

3.6.5 The IP-PABX shall support the facility where any subscriber from public network can dial the IP-PABX extension number. It can be implemented in either of the following methods:

(a) Without the assistance of any IP-PABX operator. In such scenario, on receipt of incoming call, IP-PABX shall feed second dial tone/ announcement after which the calling subscriber shall dial extension number.

(b) With the assistance of Operator- In this case, any incoming call shall be connected to operator position and the operator shall be able to transfer the call to desired extension.

3.6.6 IP-PABX Shall ensures a minimum current of 22mA on junction/trunk line. In any case, current drawn by IP-PABX shall not exceed 60 mA on junction/trunk line.

3.6.7 Line condition:

(a) The connection from IP-PABX to the main exchange shall be independent of line polarity.

(b) The IP-PABX shall be capable of detecting the calling condition (I/C call) by sensing the ringing signal (75 ± 5 V rms, 25 Hz) from main exchange.

(c) The IP-PABX shall be capable of working satisfactorily on the lines from exchange under adverse conditions of leakage i.e. insulation resistance as low as 20 kΩ between 'a' and 'b' limbs and to earth.

(d) It shall be possible to conduct the following tests from the main exchange. IP-PABX shall provide appropriate termination on its outgoing junctions, for conducting the following line testing from main exchange:

   (i) Continuity or click test as in the case of a normal telephone having capacitor 2 μF in series with the bell.

   (ii) Insulation resistance test of 'a' and 'b' limbs of the line with respect to earth and between each other. The insulation resistance shall be more than 250 kΩ during the idle conditions of the IP-PABX junctions. These tests shall be possible under all conditions of the IP-PABX such as power down, night switching etc.

(e) The IP-PABX shall provide for local battery feed to the extensions. The equipment shall also provide for DC isolation between the DC supplies used in the system and the DC battery feed from the exchange i.e. exchange battery shall not be used for IP-PABX circuitry. Either the +ve or -ve end of the local battery feed to the extensions should be connected to the general earth.

(f) Incoming junction calls to IP-PABX from main exchange that have been extended through the IP-PABX operator’s console to any extension shall either be returned back to the attendant/operator (In case of a IP-PABX not having an operator, an extension that has originally received the call shall be assumed to be the operator) or a suitable tone/announcement shall be returned to calling subscriber, if the called extension does not answer within a pre-determined time.

(g) It shall be possible to provide the facility of one way working of the junctions i.e. some of the junctions from the main exchange to be used as I/C junctions only and should not be accessed for any O/G call from the IP-PABX to the main exchange.

3.6.8 The IP-PABX shall provide for time out and release of call under held up condition during pre-dialling, dialling, post-dialling, ringing phase and Called Subscriber Held (CSH) condition.

3.6.9 The IP-PABX shall conform to the following transmission parameters for 2Wire interface of IP-PABX. The specifications for all the transmission parameters are given below. The nominal value of the impedance at the line access is 600Ω , balanced.

i) Return Loss:

The return loss measured across the nominal impedance of 600 Ω should not be less than the following values:

Frequency	Return loss
300 Hz	14 dB
400 Hz	16 dB
500 Hz	18 dB
500 Hz-2000Hz	18 dB
2700Hz	16 dB
3400Hz	14 dB

ii) Impedance unbalance about earth:

The value of longitudinal conversion loss (LCL) should exceed the minimum values given below:

Frequency	Longitudinal conversion loss (LCL)
300-600 Hz	40 dB
600-3400Hz	46 dB

iii) Cross Talk

The cross talk between individual lines should be such that with a sine wave signal of frequency at a level of 0 dBmo applied to a port, the cross talk level received in any other line should not exceed –65 dBmo.

iv) Idle State Noise

Single Frequency Noise- The level of any single frequency (in particular the sampling frequency and its multiples) measured selectively should not exceed – 50 dBmo.

3.7 PRI Interface (Applicable for Type-III Interface only)

In addition to functional requirements described in clauses 3.1 to 3.4 of this document, this clause and its sub-clauses specifies the functional requirements for

IP-PABX with TYPE-III interface (ISDN PRI i.e. 30B+D) both for outgoing calls from IP-PABX to Public network and incoming calls from the Public network.

3.7.1 The IP-PABX shall have the capability to interconnect with TDM network using ISDN Primary Rate Interface i.e. 30B+D.

3.7.2 Whenever any extension goes off-hook, the IP-PABX shall feed dial tone to its extensions.

3.7.3 Depending on the requirements of the IP-PABX, the exchange will send all/last few digits of called number to the IP-PABX. The IP-PABX shall indicate the end of digits by sending appropriate signal to the main exchange.

3.7.4 Access to Public Network: The Public Network shall be accessed by dialing single digit. Dialling of ‘0' (or any other suitable code) shall be used to provide access to the public network (O/G junctions) and the junction access should have a random or rotating priority to distribute the traffic.

3.7.5 It should be possible to transfer voice as well as data over any channel on PRI interface.

3.7.6 The IP-PABX shall provide for time out and release of outgoing junction/circuit under held up condition during pre-dialling, dialling, post-dialling, ringing phase and CSH condition.

3.7.7 Calling Line Identification

(a) For all types of calls from public network, Calling Line Identification (CLI) of a pre-defined number of digits will be sent by the main exchange. At present it is 10 digits CLI, but it can be increased up to 16 digits. The IP-PABX shall be able to receive complete Calling Line Identification (CLI), (consisting of country code + Area code + Directory number) from the main exchange (at present country code in not included in CLI). It shall not be possible to change the CLI in IP-PABX.

(b) For all types of calls made from extension to public network (If extension numbering scheme used is from ‘group of directory numbers’ allotted by service provider), CLI of extension number shall be send to the main exchange.

3.7.8 The IP-PABX shall support the facility where any Public network subscriber shall be able to dial the IP-PABX extension number without the assistance of any IP-

PABX operator/attendant. This facility can be implemented by any one of the following methods:

i. Direct Inward Dialling (DID: Service provider shall allot ‘a group of directory numbers’ from the concerned parent exchange to the IP-PABX. All extension users will have directory numbers as per E.164 numbering plan of the service provider. In such cases, main exchange shall send digits of extension number to IP-PABX.

ii. Each IP-PABX shall be allotted a single directory number by the service provider. When a Public network subscriber dials IP-PABX number, IP-PABX shall feed announcement to caller and calling subscriber shall dial the extension number

3.7.9 Alarm/Fault Diagnosis of Interface

For PRI interface, IP-PABX shall support display of status of PRI towards the main exchange and shall generate the following alarms.

a) Loss of the incoming signal at 2048 kbit/s.

b) Loss of frame alignment. – IP-PABX should detect ‘loss of frame alignment’ within 3 ms.

c) Loss of Multiframe alignment

(Multiframe alignment is assumed to have been lost when, for a period of one or two multiframes, all the bits in time slot 16 are in state 0. Multiframe alignment is assumed to have been recovered only when at least one bit in state 1 is present in the time slot 16).

d) Excessive bit error ratio.

4.0 Interface Requirements-

4.1 IP-PABX system shall interface to user terminals on one side and IP network of service provider on the other side (as shown in figure-A1)

4.2 For Extensions Side:

IP-PABX shall support SIP phone without any external IP to TDM convertor. IP-PABX shall support SIP phone with IPv4 as well as IPv6 addressing. (One at a time)

4.3 Ethernet Interface towards Public Network i.e. Type I (Applicable for Ethernet interface only)

4.3.1 IP interfaces of the system shall be configured without any external adapter (i.e. IP to TDM convertor).

4.3.2 The physical interfaces (Ethernet connections used) shall consist of 10/100 Mbps Base-T Fast Ethernet or Gigabit Ethernet over twisted pair, coaxial or Fiber (as per IEEE, 802.3u/ IEEE 802.3z). Category 5 cables/ Category 6 cable/8P8C modular connector shall be used.

4.3.3 Signalling

(a) At Layer 2, Ethernet 802.1p/Q standards define the bit markings of Ethernet packet header which are used to prioritize packets at Layer 2.

(b) At Layer 3, IP standard DiffServ defines bit markings in the Type-of-Service (TOS) fields in the IP header, which will identify a packet to be associated with a specific service. On IP equipment end-to-end, these services can be administered.

(c) Ethernet interface shall support standard based protocol (RTCP) for voice quality monitoring.

(d) Signalling between IP-PABX and IP core network shall comply the following RFCs.

S.No	Document No.	Title
1	IETF RFC 2460	IPv6 addressing
2	IETF RFC 3261	SIP: Session Initiation Protocol
3	IETF RFC 4733	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
4	IETF RFC 3550	Real Time Transport Protocol (RTP)

5	IETF RFC 793 or IETF RFC 768	Transmission Control Protocol (TCP)/ User Datagram Protocol (UDP)

4.3.4 The system shall support Network Time Protocol as per RFC 1305 to synchronies the system date and time with network devices.

4.4 2W-Interface i.e. Type II (Applicable for 2W interface only)

4.4.1 For outgoing calls from IP-PABX to the main exchange, IP-PABX shall use DTMF signalling. Applicant may apply for decadic signalling also. If applicant applies for decadic signalling, same shall be indicated on the TAC, after testing.

4.4.2 Decadic Signalling - In case of decadic dialling out by IP-PABX on the junctions, decadic pulses shall have the following characteristics (Applicable only if the applicant has applied for decadic signalling):

i Dial speed shall be 10 ± 0.5 IPS.

ii Make/break ratio shall be 1:2 nominal with break period between 60% to 70%

iii IDP of 800 msec. ± 10%

4.4.3 DTMF signaling –

(a) In case of DTMF dialling out by the IP-PABX, the specifications for the same shall be in conformity with ITU-T recommendation Q.23. The level of the signals shall be as follows:

Low group	-8 ± 2 dBm
High-group	-6 ± 2 dBm
Pre-emphasis	2 ± 1 dB

(b) Transmission of DTMF Signalling shall be as per ITU-T recommendation Q.23 and reception of DTMF Signalling shall be as per ITU-T recommendation Q.24. The number of digits/characters used for Signalling shall be 12 and a set of two frequencies shall be used for each digit/character.

Low frequency group : 697,770, 852, 941 Hz

High frequency group : 1209, 1336, 1477 Hz

Digit	Frequencies used	Digit	Frequencies used
Digit 1	697 & 1209	digit 7	852 & 1209
digit 2	697 & 1336	digit 8	852 & 1336
digit 3	697 & 1477	digit 9	852 & 1477
digit 4	770 & 1209	digit 0	941 & 1336
digit 5	770 & 1336	*	941 & 1209
digit 6	770 & 1477	#	941 & 1477

(c) Each transmitted Signalling shall be within ± 1.8% of the nominal frequency.

(d) The total distortion products (resulting from harmonic or inter-modulation) must be at least 20 dB below normal frequencies.

(e) Level difference between any two frequencies forming a signal should not be more than 2 dB ± 1 dB such that the higher frequency component of the signal is of higher level.

(f) During DTMF Signalling, the D.C. loop shall be maintained.

(g) The IP-PABX shall not provide for more than one automatic repeat dialling on the junction line. In case of DTMF redial, the on/off period shall be at least 40/40 msecs.

4.5 PRI Interface (Applicable for PRI Interface only)

4.5.1 Layer 1 requirements shall be as per ITU-T- Primary Rate User-Network Interface – Layer 1 Specification.

The primary rate shall support only the point-to-point configuration. Point-to-point configuration at layer 1 implies that for each direction only one source (transmitter) and one sink (receiver) are connected to the interface.

i). Nominal bit rate: 2048 kbit/s.

ii). Bit rate accuracy: ±50 ppm (±102.4 bit/s).

iii). Code: High density bipolar of order 3 (HDB3)

4.5.2 Layer 2 requirements shall be as per ITU-T Q.920 - ISDN user-network interface data link layer-General aspects and Q.921 - ISDN user-network interface - Data link layer specifications

4.5.3 Layer 3 message structure and coding of information elements shall be as per ITU-T Recommendation Q.931.

5.0 Quality Requirements:

5.1 Codec support (for Ethernet Interface only)

5.1.1 Ethernet interface shall support the procedures for Codec negotiation, in association with the Main switching Node. It shall support at least the following codec's as per latest ITU-T recommendations

	Type of service	Codec to be supported
a)	Voice Call	G.711(μlaw) G.729 A
b)	FAX	T.38
c)	Video applications	a) H.264

Applicant may apply for additional codec supported and the same may be mentioned in the approval certificate after testing

5.1.2 It shall be possible to convey the 'list of available codec that can be used for the call set-up during active phase of the call

5.1.3 The Codec modification procedures shall also be provided, wherein the Codec selected for a call can be modified in any direction and at any time during the active phase of the call.

5.1.4 Any voice processing functions shall not interfere with transparent interchange of FAX signals.

5.1.5 It shall automatically recognize the voice, FAX & data traffic coming from Public network and shall do the required compression.

5.1.6 It shall generate voice quality deterioration alarms in case RTP characteristics go down below a configured threshold. It shall not contribute to Packet Loss. The one way Delay introduced by the IP-PABX for different encoding schemes shall be as per Table I.3 of ITU-T Rec G.114.

5.1.7 Echo Cancellation: It shall support integrated echo cancellations, as per ITU-T recommendations G.168 up to a configurable value of 128 ms. It shall be configurable to activate and deactivate the echo cancellation on each trunk group. Activation of Echo cancellation shall not result in decrease in number of circuits.

5.2 In case of Ethernet interface, IP-PABX should support RTCP-XR protocol for monitoring real time VoIP voice quality (For calls from IP to analogue & vice-versa).

6.0 EMI/EMC Requirements

6.1 Electromagnetic Interference

The equipment shall conform to the following EMC requirements for Class A:

Electromagnetic Compatibility (EMC) Requirements:

The equipment shall conform to the EMC requirements as per the following standards and limits indicated therein. A test certificate and test report shall be furnished from an accredited test laboratory.

a) Conducted and radiated emission (applicable to telecom equipment):

Name of EMC Standard: “CISPR 22 {2005} with amendment 1 (2005) & amendment 2 (2006) - Limits and methods of measurement of radio disturbance characteristics of Information Technology Equipment".

Limits:-

i) To comply with Class A of CISPR 22 {2005} with amendment 1 (2005) & amendment 2 (2006)

ii) The values of limits shall be as per TEC Standard No. TEC/EMI/TEL-001/01/FEB-09.

b) Immunity to Electrostatic discharge:

Name of EMC Standard: IEC 61000-4-2 {2001) "Testing and measurement techniques of Electrostatic discharge immunity test".

Limits: -

a. Contact discharge level 2 {± 4 kV} or higher voltage;

b. Air discharge level 3 {± 8 kV} or higher voltage;

c) Immunity to radiated RF:

Name of EMC Standard: IEC 61000-4-3 (2006) "Testing and measurement techniques-Radiated RF Electromagnetic Field Immunity test"

Limits:

i) Under Test level 2 {Test field strength of 3 V/m} for general purposes in frequency range 80 MHz to 1000 MHz and

ii) Under test level 3 (10 V/m) for protection against digital radio telephones and other RF devices in frequency ranges 800 MHz to 960 MHz and 1.4 GHz to 6.0 GHz.

d) Immunity to fast transients (burst):

Name of EMC Standard: IEC 61000-4-4 {2004) "Testing and measurement techniques of electrical fast transients/burst immunity test"

Limits:-

Test Level 2 i.e. a) 1 kV for AC/DC power lines; b) 0. 5 kV for signal / control / data / telecom lines;

e) Immunity to surges:

Name of EMC Standard: IEC 61000-4-5 (2005) “Testing & Measurement techniques for Surge immunity test"

Limits:-

i) For mains power input ports : (a)1.0 kV peak open circuit voltage for line to ground coupling (b) 0.5 kV peak open circuit voltage for line to line coupling

ii) For telecom ports : (a) 0.5 kV peak open circuit voltage for line to ground (b) 0.5 KV peak open circuit voltage for line to line coupling.

f) Immunity to conducted disturbance induced by Radio frequency fields:

Name of EMC Standard: IEC 61000-4-6 (2003) with amendment 1 (2004) & amendment. 2 (2006) "Testing & measurement techniques-Immunity to conducted disturbances induced by radio- frequency fields"

Limits:-

Under the test level 2 {3 V r.m.s.} in the frequency range 150 kHz-80 MHz for AC / DC lines and Signal /Control/telecom lines.

g) Immunity to voltage dips & short interruptions (applicable to only ac mains power input ports, if any):

Name of EMC Standard: IEC 61000-4-11 (2004) “Testing & measurement techniques-voltage dips, short interruptions and voltage variations immunity tests”

Limits:-

i) a voltage dip corresponding to a reduction of the supply voltage of 30% for 500ms (i.e. 70 % supply voltage for 500 ms)

ii) a voltage dip corresponding to a reduction of the supply voltage of 60% for 200ms; (i.e. 40% supply voltage for 200ms) and

iii) a voltage interruption corresponding to a reduction of supply voltage of > 95% for 5s.

Note1:For checking compliance with the above EMC requirements, the method of measurements shall be in accordance with TEC Standard No. TEC/EMI/TEL-

001/01/FEB-09 and the references mentioned therein unless otherwise specified specifically. Alternatively, corresponding relevant Euro Norms of the above IEC/CISPR standards are also acceptable subject to the condition that frequency range and test level are met as per above mentioned sub clauses (a) to (g) and TEC Standard No. TEC/EMI/TEL-001/01/FEB-09.The details of IEC/CISPR and their corresponding Euro Norms are as follows:

IEC/CISPR	Euro Norm
CISPR 11	EN 55011
CISPR 22	EN 55022
IEC 61000-4-2	EN 61000-4-2
IEC 61000-4-3	EN 61000-4-3
IEC 61000-4-4	EN 61000-4-4
IEC 61000-4-5	EN 61000-4-5
IEC 61000-4-6	EN 61000-4-6
IEC 61000-4-11	EN 61000-4-11

The manufacturer/supplier shall submit a test report for EMI/EMC compliance. The test agency for EMI/EMC compliance shall be a credited agency and details of accreditation shall be submitted.

7.0 Safety Requirements

7.1 The equipment shall conform to IS 13252 part 1:2010- "Information Technology Equipment – Safety- Part 1: General Requirements" [equivalent to IEC 60950-1 {2005}] "Information Technology Equipment –Safety- Part 1: General Requirements" and

7.2 A test certificate and test report shall be furnished from a test agency.

7.3 The test agency for safety requirements tests shall be an ISO 17025 accredited agency and details of accreditation shall be submitted.

8.0 Security Requirements

8.1 Password Management

8.1.1 Access to system operations shall be controlled through multi-level password and authentication checks

8.1.2 The man-machine communication programs shall have the facility of restricting the use of certain commands or procedures to certain passwords and terminals

8.1.3 It shall be possible to define users/user groups with different access rights

8.1.4 It shall be possible to modify user password number of times.

8.1.5 Session ID shall be logged with information of user ID, password, time of login, commands/parameters given etc.

8.1.6 All passwords shall be stored in encrypted form and no user including 'Network Manager' shall be able to read the password.

8.1.7 The system must support 'session logout timing with configurable time periods

8.1.8 The system should block the access from local as well from remote terminals after receipt of consecutive predefined (say 5) wrong login/passwords and unauthorised commands.

8.1.9 The system must support password encryption, usage of AES (Advanced Encryption System)-128 Bit algorithm for password encryption and shall be kept in # form

8.2 System shall be configured in such a way that it does not support direct, externally initiated, connections via HTTP, telnet, FTP, TFTP or any other protocol as means to prevent distributed Denial of Service attack exploitation (except for management purpose).

8.3 Operator/Maintenance console shall be GUI based. Necessary software package to prevent loading of any unauthorised software or driver on the I/O terminal shall be provided.

8.4 All management traffic (including user information) between the remote console/session and IP-PABX server shall be encrypted. (SSH for Direct Command Line Session, Interface, https (SSL) for Web sessions, SFTP for file transfer etc.)

8.5 The protection of signaling connections over IP by means of authentication, Integrity and encryption shall be carried out using TLS (Transport Layer Security Protocol)

8.6 It shall be possible to connect IP-PABX through network separation methods such as SBC, firewalls, virtual router (sub-netting) at the edge of service providers network.

9.0 Various requirements of the category/configuration of the product for testing

In addition to Ethernet interface, applicant may apply any one or both type(s) of interfaces such as

a. 2W DEL
b. PRI Interface

CHAPTER-2

10 **Desirable Requirements**

The various functions/facilities/ features described in this chapter are comprehensive and suggestive which may be useful. These need not be treated as mandatory for the product. However, the purchaser will select the functions/ facilities/features of IP-PABX as per its requiremets at the time of procurement/ tendering.

The functions/ facilities/ features described in this chapter will not be tested/ verified by TEC

10.1 General

10.1.1 The system architecture may be so designed that it enables expansion/ up gradation of the system without any compromise with existing features/ functionality.

10.1.2 The system software may be modular so that any functionality can added or removed without disturbing the other functionalities.

10.1.3 System may support MLPP feature.

10.2 System Redundancy:

10.2.1 The System redundancy (Optional): The system may support duplicated Call server in Hot standby mode.

10.2.2 Following units may be provided with redundancy

a) MGW
b) All types of Interface cards
c) Power supply

10.2.3 The standby server should automatically synchronize with active server and should take over the database and telephony functions seamlessly in case of failure of main active server without need of manual configuration & administration.

10.2.4 It should be possible to reach the ultimate capacity of the IP-PABX without any degradation of IP-PABX system.

10.2.5 The system should adopt active/standby mode, load sharing and redundancy configuration for the servers and optimizes fault detection and isolation techniques of the faults.

10.2.6 Authorization Code: Every user may have own authorization code to make outgoing calls thereby ensuring no misuse of the system. System may give the user complete flexibility to dial his Personal Identification number/ authorization code from any location by dialing authorization code and may be able to use all his facilities. All call made from other location by dialing authorization code may be billed against authorization code and not against extension number.

10.3 System Security:-

a. The System may support Syslog services for command and configuration control accounting with a minimum of 5 day history.
b. Internal OS controls for remote point of access restriction and service availability
c. Account access authentication/restriction at different levels may be provided so as to prevent unauthorized access or interference to services, calls, protocol and data.

10.3.1 IP-PABX may support the deployment of network separation methods such as SBC, firewalls, virtual routers (sub-netting) at the edge of IP-PABX. (Applicable only in case of Ethernet interface)

10.3.2 IP-PABX shall support the Diffserv as per RFC No. 2474, 2475. IP-PABX shall be able to set the Type of Service (TOS) bits depending upon the Codec, priority subscribers, incoming PSTN Ports etc.

10.3.3 Voice Activity Detection (VAD) and silence suppression functionality shall be integrated with voice codec. It shall be possible to activate and deactivate VAD and Silence Suppression. The Trunk Media Gateway shall support the Comfort Noise Generation and insertion. This shall not result in performance deterioration.

10.4 Maintenance Console:

10.4.1 The offered system may be provided with a PC based and software up-gradable maintenance console.

10.4.2 System administration may be password protected with notification of security violations.

10.4.3 System may be capable of maintenance facility from a central location over LAN within the enterprises.

10.4.4 The system management may enable administrator to navigate, display, add modify and/or remove the system and related switch components. The system management may enable administrator to allocating, changing and removing any fuser's facility on any extension number.

10.4.5 The system may support fault diagnosis.

10.4.6 Call statistic reports as required may be available on the monitor as well as printouts. The formats of printouts may be programmable

10.4.7 It shall be possible to implement and provision services, based on following capability types, for which APIs shall be provided:

- Identification
- Authentication
- Authorization
- Location
- Presence and Availability
- Group management (CUG/VPN)
- Call Control :
    - Generic
    - Multi-Party
    - Multimedia
    - Conference
- Session Control
- Service Subscription (Registration) Management
- User and Terminal Profile Management
- Generic messaging
- Multimedia messaging
- Push
- Content based charging
- Emergency Communication Management
- Address list management
- Prioritized communication /traffic handling
- Service Independence - service capabilities which will allow IP multimedia applications to be deployed in a vender independent manner.
- Application Service inter working - Allows inter working of application service protocols. This capability may allow inter working between application services and network entities for execution of the service

10.4.8 It may provide the capabilities to:

- Determine the location of the target end point.

- Determine the media capabilities of the target end point via Session Description Protocol (SDP).
- Determine the availability of the target end point.
- Establish a session between the originating and target end point if the call can be completed.
- Handle the transfer and termination of calls. SIP supports the transfer of calls from one end point to another

10.5 Auto-attendant: This is the part of the system that initially answers the incoming calls, eliminating the need for a full-time receptionist. Callers hear a custom business greeting, and then are given options for routing their calls.Various option for routing the call may be configurable.

10.6 Operator Console

Operator consol may support the following features:

i The Operator may be able to identify the category of an incoming call e.g. internal, external, diverted, networked

ii The operator may be able to transfer incoming calls to another number, internal or external.

iii. The operator may be able to camp on a call on to a busy extension.

iv. The operator may be able to make an internal/external call and may be able to transfer it another number.

v. The operator may be able to park calls and retrieve them later.

vi. An extension may be able to dial the operator by a general access code (say 9) and by an individual extension number.

vii. Busy Lamp Field – to indicate the busy/ free status of the defined extensions and trunks groups.

10.7 PC for Operator Console & Maintenance Console-Following are the suggestive specifications for PC for Operator/ maintenance console:

a. Windows XP or higher with Ms office 2010 professional or Linux/IOS
b. Intel Pentium 4 Processor, 2.9 GHz or above
c. HDD – 100 GB
d. RAM - 512 MB DDR RAM (upgradeable up to 1 GB)
e. Ethernet Interface (10/100/1000Mbps)
f. USB Ports
g. FDD -1.44’

h. 52 X CD ROM/DVD with Multimedia

i. Keyboard

j. Mouse

k. Flat Colour Monitor (Minimum size 17”)

10.8 Diagnostic programs to localise faults: On a faulty condition, the software may provide for locating the faulty subsystem.

10.9 Power Supply: Purchase may specify any one of the following option for power supply

Option 1: The equipment may be capable of working with –40 V to –57 V. D .C. input from power supply. Switching mode Power Supply (SMPS) battery Power supply and battery may be modular and expandable to support the ultimate equipment configuration.

Option 2: AC Mains supply of 220 Volts with a tolerance of -15% to + 10% would be available. The frequency may be 50 Hz $\pm$ 2 Hz.

Purchase may specify the power requirement as per option 1 or 2.

10.10 Day and night modes- IP PABX system acts differently after business hours. This feature allows the system to route incoming calls depending on the current time of day. One may have several after-hours options, available at the click of a button:

(a) Change the greeting callers hear, but leave the system fully active. Callers will know that business hours are over, but may still try to connect to an extension. With follow-me calling, answering options and voicemail each extension can decide if he wants to continue to take calls or simply collect messages.

(b) Change the greeting and send all calls to voicemail. Called umber may know that messages have been recorded through notification features.

(c) Leave the system in full operation, or put it back when the office opens again.

10.11 User Facilities

IP-PABX may provide the following facilities/services for all types of extensions:

10.11.1 O/G Call Restriction – Administrative controlled

10.11.2 O/G Call Restriction – Subscriber controlled

10.11.3 Automatic Alarm Call service (Wake-up-call)

10.11.4 Call Waiting

10.11.5 Absent subscriber Service:

10.11.6 Do Not Disturb

10.11.7 Call Forwarding on Busy, No Reply, Immediate

10.11.8 Incoming only line

10.11.9 Outgoing only lines

10.11.10 Audio Conference Call:

10.11.11 Fixed Destination Call - Immediate or Timed out (Hot line)

10.11.12 Call Transfer

10.11.13 Call Forwarding to Fixed Number

10.11.14 Automatic Call Back

10.11.15 Call Completion on Busy Subscriber (CCBS)

10.11.16 Call Monitoring

10.11.17 Call Parking

10.11.18 Call Pick-up

10.11.19 Call Queuing.

10.11.20 Call Recording

10.11.21 Call Blocking

10.11.22 Call ID on Call Waiting

10.11.23 Distinctive Ringing

10.11.24 Music on Hold

10.11.25 Music on Transfer

10.11.26 Custom greeting

10.11.27 Personal greetings

10.11.28 Call Return / Camping

10.11.29 Follow-me / Find-me

10.11.30 Message Waiting Indicator (MWI)

10.11.31 Voice mail or VMS

10.11.32 Closed User Group:

10.11.33 Parallel Ringing

10.11.34 Conference Bridging

10.11.35 Dial by Name

10.11.36 Local and Remote Call Agents

10.11.37 Automated directory

10.11.38 Call screening

10.11.39 Soft Phone Features: Following Soft Phone features may be supported:

i Multi-Location Functionality: User may be allowed full access to all the features of the system from any user terminal.

ii Voicemail via Email: IP-PABX sends digital voicemail sound files and Caller ID via email. It may be possible to scan, forward and organize voicemail messages. This may be in addition to traditional voicemail retrieval (dialing a voicemail box).

iii Call Record: It may be possible to make MP3/WAV recordings of phone conversations, simply by pressing the "Record" button on IP phone. Recordings may be sent automatically o user’s email box after the call ends.

10.11.40 Selective Call Forward

10.11.41 Selective Call Rejection

10.11.42 Instant messaging (IM) and presence:

Presence information and notification may be provided as below:

i Information that client is registered

ii Information that Client is currently engaged in an Instant Messaging Session.

iii SIP clients may communicate with SIP Server using SIMPLE protocol.

iv Click-to-Call - Initiate a call from any directory entry

10.11.43 IP terminal may provide the capabilities to:

- Determine the location of the target end point.
- Determine the media capabilities of the target end point via Session Description Protocol (SDP).
- Determine the availability of the target end point.
- Establish a session between the originating and target end point if the call can be completed.
- Handle the transfer and termination of calls. SIP supports the transfer of calls from one end point to another

10.12 Suggestive specifications of SIP hard phone:

SIP hard phone may have least he following features:

i Alphanumeric Keypad, programmable function key with LED indicator.

ii At least 2-line 7’ LCD alphanumeric display with backlight

iii Camera 12 Pixels

iv Video format mp4

v Audio format mp3

vi Picture format jpeg, png

Purchaser shall specify the additional features, if any, as per his requirements.

10.12.1 Suggested Video Resolution:

System may support minimum video resolution of

a Pro-Motion interlaced video (60/50 fields full-screen video for NTSC/PAL)

b 4SIF (704 x 480)

c 4CIF (704 x 576)

d SIF (352 x 240)

e CIF (352 x 288)

10.12.2 IP-PABX may comply to following Protocols/ specifications for SIP terminals:

An Offer/Answer Model with the Session Description Protocol (SDP), RFC 3264

10.12.3 Following RFC may be supported: 2246, 2543, 3266, 3312, 3323, 3326, 3420, 3428, 3455, 3515, 3556, 3588, 3665, 3725, 3856, 3863, 3880, 3891, 3911, 3966, 4028

10.12.4 SIP Phones may support 802.1x (EAP-MD5/AES or better) for authentication and access control to the network i.e. user may be connected to network only after he has passed the authentication process

10.12.5 SIP Phone may support built in Ethernet switch to cascade PC with the phone so that single I/O port is used to connect both SIP phone and PC

10.13 Analogue Telephone Instrument specifications

Analogue type of phone may have minimum following features:-

i The telephone may be provided with Display screen which have the following facility:

a Display of called number

b Visual indication of incoming call

c Duration of successful call

d Call waiting indication

e Real time clock

f Number storage (at it may be possible to store 50 numbers)

g CLI display (up to 16 digits)

h Loudspeaker (hands free operation)

ii In addition loudspeaker (hands free operation), Manual pause feature may also be provided.

iii Dual-tone ringer with volume adjustable in 3 steps (high, medium & low) may be provided

iv The telephone instrument may not be damaged when intermittent voltage pulses each of up to 250 ms duration, between earth wire and the line wires connected together, of up to 650 V at 50 Hz are applied.

v It may be able to support DTMF signalling.

10.14 Digital Extension

In addition to user facilities described above in clause 10.11, IP-PABX system may support the following bearer services, tele-services and supplementary services on digital extensions:

10.14.1 The system may support the following Number Identification supplementary services: as defined in ITU-T Recommendations I.251, Q.81 and Q951.

10.14.2 Multiple Subscriber Number (MSN):

i As up to 8 terminals can be connected in parallel on the subscriber premises wiring, to call a specific terminal (PC to call a PC, and phone to call a phone) separate number may be allotted to each terminal. It may be possible to program each MSN no.

ii The Customer Premise Equipment (CPE) with the MSN supplementary services may use the least significant “n” digits up to the total number of digits supplied as part of selection process. If a user receives fewer digits than it requires for terminal selection then the user may use the available information in the called party number information element for its terminal selection.

iii Assigning bearer services, tele-services and supplementary services and outgoing call barring MSN wise may be possible.

10.14.3 Multiple calls: It may be possible to have two voices, fax or PC "conversations," and one data "conversation" at the same time, through the same ISDN connection.

10.14.4 D-channel messages may contain signalling data (s-data) as well as packet oriented user data (p-data), provided that the appropriate terminal equipment for packet-oriented data transmission is available where as the Primary Access D-channel may only be used for signalling information (i.e. for s-data). Packet-oriented user data (p-data) must be separated from the s-data in the IP-PABX and transferred to D-channel. Using the basic channels, transfer of data may be possible during conversation without disturbing the voice calls.

10.14.5 CLIP: When a digital subscriber receives a call, the calling subscriber number is displayed on digital telephone before the called subscriber answers the call. Thus, the called subscriber knows the telephone number of calling party even before answering the call. For example when the subscriber is already in conversation, he may choose to attend the second incoming call depending on the caller’s number displayed. Calling line identification presentation may be available to all the digital subscribers by the default

10.14.6 CLIR: The calling subscriber with the facility of CLIR will be able to prevent the presentation of his number to the called subscriber (Prevention of CLIP). However, this service will be over ridden by certain agencies such as police and fire services, since they may need to know the identity of the caller in all cases.

10.14.7 COLP: When the called subscriber answers the call, MSN number of called terminal may be presented on calling digital telephone

10.14.8 COLR: The calling subscriber with the facility of COLR will be able to prevent the presentation of his MSN number to the called subscriber (Prevention of COLP)

10.14.9 Sub-addressing: It may be possible to assign separate sub-address to each terminal

10.14.10 Call Offering supplementary services:

The system may support the following Call offering supplementary services as defined in ITU-T Recommendations I.252, Q.82 and Q952. Coding of Redirecting number information may be as per ITU-T recommendation Q.952

10.14.11 CFB (Call forwarding on busy) - If the called subscriber is busy, the incoming calls to his number may be diverted to another number specified by him.

10.14.12 CFNR (Call Forwarding No Reply) - If the called subscriber is not available (or does not answer the call), the incoming calls to his number may be diverted to another number specified by him after a few rings.

10.14.13 CFU (Call Forwarding Unconditional) - All the incoming calls to subscriber may be diverted to another number specified by him. In this case the ring directly goes to diverted number

10.14.14 Call Completion supplementary services:-The system may support the following Call completion supplementary services as defined in ITU-T Recommendations I.253, Q.83 and Q953.

10.14.15 Call Waiting (CW) - When any subscriber attempts to make a call to subscriber who is already engaged in another call second incoming call is announced via beeps. The arrival of second call may be displayed as the call waiting message as well as is announced via beeps and by pressing the call waiting button. It may be possible to take the second call.

10.14.16 Call Hold (CH) - An external call can be put on Hold on any of the extensions while a second call is to be made or answered. The subscriber can switch between these calls.

10.14.17 Multiple Supplementary Services:

The system may support the following Multiple supplementary services as defined in ITU-T Recommendations I.254, Q.84 and Q954

10.14.18 3PTY (Three party conference): This enables the user which has one held call and one ongoing call in conversation to joins the held up call also to form a three party conference i.e. all the 3 party are in conversation. It may be possible to have 3 way conference call with either two external numbers and one internal extension or one external number and two internal extensions.

10.14.19 Additional Information transfer supplementary services- The following Additional Information transfer supplementary services may be supported as per ITU-T recommendations I.257, Q.87 & Q.957

10.14.20 UUS: User-to-User signalling supplementary service 1, 2 & 3 (UUS1, UUS2 & UUS3) using implicit procedure may be supported. It may be possible to transfer up to 128 octets of user information in each message.

10.14.21 Mobility and Modification supplementary services: The following Mobility and Modification supplementary services may be supported as per ITU-T recommendations I.258, Q.83 & Q.953.

10.14.22 TP (Terminal Portability) - In the subscriber premises up to 8 terminals can be connected to a single digital line. These terminals can be in different rooms and also can be on different floors. During conversation it is possible to transfer the call from one terminal to another or even remove the terminal and connect it to another socket at a different location. This facility may be available for calling as well as called subscriber.

10.14.23 CT(Call Transfer) - Call can be transferred to any extensions, while first call is in progress

10.14.24 CD (Call Deflection) - It may be possible to divert the call in ringing phase

10.14.25 CONF (Conference Call) - It may be possible to introduce a common speech path so that all the subscribers in conference are able to speak to each other. Purchaser may specify the maximum numbers of subscribers in a conference and the maximum numbers of such simultaneous multi-party conferences

10.14.26 AOC–E (Advise of charge at the end of call) - It may be possible send charge information to user side at the end of call. The amount charged for a call, in terms of call units, may be displayed on the LCD of calling subscriber's digital phone.

10.14.27 AOC–D (Advise of charge during the call) - It may be possible send charge information to user side during conversation. The amount charged for a call, in terms of call units, may be displayed on the LCD of calling subscriber's digital phone

10.14.28 Speed Calling: It may be possible to store and dial some frequently used numbers (local or long distance) by dialling only two digits

10.14.29 ECT (Explicit Call Transfer) - This will enable the user to connect two calls into one call and may be able withdraw from the call

10.14.30 En-block Dialling – IP-PABX may support en-block dialling from digital phone

10.14.31 Overlap Dialling - IP-PABX may support overlap dialling from digital phone

10.15 Digital Telephone Instrument:

Digital Phones may have at least the following features:

- i Tilt able Display 40 alphanumeric characters display
- ii Volume control Keys-Increase and Decrease
- iii Programmable ringer, loudness and tone character
- iv Feature keys. - Inclusive of Mute, Hands free and Headset function keys, Enquiry key, Transfer key etc.
- v Message waiting indicator LED
- vi Fully duplex
- vii Menu driven function for feature access
- viii Menu Navigating Keys

ix Min 12 Programmable Keys

x Adapter position for Digital/ Analog Phone/ digital phone Interface

xi Support for Master slave configuration

xii Name dialling

xiii Alpha numeric keypads

xiv Automatic Call back

xv The subscriber unit may be powered from the system itself (over the same subscriber line pair).

xvi The display must show at least the following information:

a Time and date

b Own telephone number

c Telephone status

d Incoming/outgoing call numbers i.e. caller name/number display

e Duration of call in case of outgoing successful call

10.16 Ethernet Interface:

Signalling between IP-PABX and IP core network shall comply the following RFCs.

S.No	Document No.	Title
1	IETF RFC 3262	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
2	IETF RFC 3263	Session Initiation Protocol (SIP): Locating SIP Servers
3	IETF RFC 3389	Real-Time Transport Protocol (RTP) Payload for Comfort Noise (CN)
4	IETF RFC 2327	SDP: Session Description Protocol
5	IETF RFC 1034	Domain names – Concepts and Facilities
6	IETF RFC 2782	A DNS RR for specifying the location of services (DNS SRV)
7	IETF RFC 2915	The Naming Authority Pointer (NAPTR) DNS Resource Record
8	IETF RFC 2916	E.164 number and DNS
9	IETF RFC 2326	Real Time Streaming Protocol (RTSP)
10	IETF RFC 3311	The Session Initiation Protocol (SIP) UPDATE Method

11	IETF RFC 2806	URL'S (Uniform Resource Locator)for Telephone calls
12	IETF RFC 3551	Real Time Transport Protocol (RTCP)
13	IETF RFC 3265	Session initiation Protocol (SIP) – Specific Event Notification
14	IETF RFC 3761	E.164 Uniform Resource Identifier(URI) Dynamic Delegation Discovery System (DDDS) Application (ENUM)

10.16.1 Signaling between IP-PABX system and SIP Phone shall be encrypted using TLS (Transport Layer Security) Protocol.

10.16.2 System shall support SRTP (Secure Real-time Transport Protocol) as per IETF RFC 3711 to provide confidentiality, message authentication, and reply protection to the control traffic for RTP, the Real-time Transport Control Protocol (RTCP).

10.16.3 QoS management

i. The interface may support QoS marking and mapping as well as Priority marking and mapping

ii. The interface may support re-map of ToS bits and DiffServ code points between networks for QoS enforcement.

iii. The interface may support policy based admission control features on the basis of call details, QoS and bandwidth.

iv. The interface may support flow reporting which includes functions such as Quality and Service Level Agreement (SLA) monitoring.

10.16.4 Traffic Measurement and Recording

i. Traffic monitoring and shaping feature may be provided. Traffic report may be generated for I/C and O/G traffic.

ii. The following measurements may be provided:

a. Processors occupancy.

b. Total traffic handled in Mbps.

c. Total traffic carried by the Ethernet Interface. The total RTP traffic carried by the Ethernet Interface. Rate, ratio and total number of error packets.

d. Number of Ethernet frames sent out. .

e. Number of re-transmitted control/signalling messages during a period.

f. RTCP traffic handled.

g. Traffic handled per codec type.

iii. It may be possible to activate and record/print the measurement of delay, packet loss per session also.

iv. The above measurements, may not unduly affect the call handling capacity of the processor.

10.16.5 Performance of Operators: The offered management platform shall indicate the status regarding the Number of calls assigned to the operator and number of calls cleared and number of calls dropped etc.

10.16.6 The system may support Wireless SIP hard phones. These phones get controlled by the PBX for telephony features and gets physical connectivity using 802.11a/b/g interface from underlying Wi-Fi network

10.17 Other requirement of IP Extensions/Multi-Media PC

- IP-PABX may provide interface for connecting “SIP Telephones or PC based Video Phone with suitable Client software” which may work as extensions.
- IP Terminal terminals may be connected to IP-PABX over Ethernet as per IEEE 802.3

10.17.1 Following additional functions may be provided on IP soft phone-

a) Easy, web-based extension configuration: While overall system configuration shall be done by a system administrator, many functions and features are controlled directly by each extension owner. This gives flexibility to the employee, while reducing the load on the administrator. Typical configuration activities by extension owners include establishing follow-me lists, setting voicemail and fax forward options, checking voice and fax messages, and logging into ACD queues etc. The web configuration tool may be straightforward and easy to use. It shall be possible to do most subscriber functions from any touch-tone telephone

b) Call-answer security: With follow-me calling, one can send his calls to any phone. Some of these phones might be used by other people. Call-answer security lets user set a password that must be entered before a call is actually connected. If the password is not given, the call will move to the next number in the follow-me list, or to voicemail

c) Never full: Systems shall have dynamic voicemail storage. System shall automatically open more message space for the user when his space gets full, moving older messages out to make room for newer messages. The older messages are kept for pre-specified period, giving you ample time to retrieve them.

d) Centralized messages: Some user may have several phones, each with its own voicemail. It shall be possible to centralize all the messages in one place. User don't have to search through multiple systems, he/she just look in one place.

e) Notifications: It shall be possible to notify the user receipt of any voice message or fax for him. Notification can be email notification, pager signals, web alerts and text messaging on his/her cell phone.

f) Fax viewing options: IP-PABX shall handle incoming faxes automatically, without the need for a fax machine. The faxes shall be delivered as .pdf files, and can be viewed through web tool or can be forwarded to user email for viewing.

g) Direct Inward System Access (DISA): The Direct Inward System Access feature allows a calling party to access internal features of the system from an outside telephone line.

h) Video communication: With respective cameras installed, a terminal user can directly originate a video call to the opposite party and the appropriate video quality can be determined depending on the network bandwidth.

i) File transfer: Transfers files to another user or department where the received files can be saved under a particular directory or a specified directory.

j) Application share: A terminal user shares an application; then the opposite party can use this shared application remotely.

k) Electronic whiteboard: Both parties can write and draw on the same picture, for example, for discussion purposes. This is applicable to many occasions such as remote teaching and technical exchange.

l) Content release: The contents of advertisements and media streams can be released through a multimedia terminal, and a platform can be used to selectively locate the user or to immediately release.

m) Instant messaging: Allows real-time communication by means of text between one terminal user and another who has already logged in

n) Customized Phone Profile: It shall support web based customizable skin templates. Such functionality allows user to update the underlying image files and create customized soft phones. Additionally, each provider can create multiple phone profiles with different design, language, and functionality and use them to target different customer groups.

10.18 Capacities of the IP-PABX

Following parameters required for dimensioning purpose may be specified by the tendering authority:

i. Number of extensions of each type
ii. Maximum Number of client Registration
iii. Registration rate/registration time
iv. Number of calls/sec
v. Number of concurrent calls
vi. Call Holding time per subscriber and operator.
vii. Percentage Mix of Intra-PABX calls and calls to/from public network
viii. Other Optional parameters such as Number of trans-coded calls
ix. Optional functions and user applications, if required

10.18.1 Purchaser may define the following parameters of IP-PABX server, as per his requirements at the time of procurement-

a. Microprocessor
b. Memory
c. Hard disk
d. USB ports
e. Ethernet ports
f. Operating System

10.18.2 Type of interface(s) towards public network and number of interfaces of each type. Communication Protocols may also be specified

10.18.3 Type of Ethernet backbone (e.g. 10 Mbps-10Base-T Ethernet, 100 Mbps-Fast Ethernet or 1000 Mbps) may be indicated.

10.18.4 Various dimensioning parameters e.g. traffic handling capacity, storage type and volume, synchronous or asynchronous etc. may be indicated.

10.18.5 If any additional Codec support is required, same may be specified. Some suggested codec's are AMR-FR, AMR-HR, EFR, HR (for GSM application), EVRC (for CDMA application), H.263 (For Video application), and NNB-OPUS (For voice call).

10.18.6 Optional functions and user applications, if required

10.18.7 Total CDR & Command log storage capacity required to meet the requirements as per prevailing rules/regulations/law of Govt. of India.

10.18.8 Details of hardware, software and number of operator terminals for operation, administration and maintenance may be specified.

10.18.9 Tools and Testers required, if any

10.18.10 The period for which the maintenance spares are required, may be specified by the tendering authority.

10.18.11 Battery Requirements. Power requirements of AC and DC power supplies. Number of sets to be supplied

10.18.12 Documentation: Number of copies (hard/soft) to be supplied.

10.18.13 Qualitative Requirements (QR): The purchase shall specify quality standards like ISO 9002 or ISO 9001: 2000 certification.

10.18.14 Environment Conditions: The purchaser shall specify the requirements of Environment Conditions that the system may satisfy as specified in Quality Measure Manual for relevant category of equipment.

10.19 Some suggested configurations

S.No.	Configuration	Total No. of extensions		No. of ports in extension in Basic configuration *		No. of Registrations/sec	No. of concurrent Calls		No. of 2W junctions	No. of PRI	No. of Console
		Basic Capacity	Expandable up to	SIP	Analogue		IP to IP	IP to TDM			
1.	A	8	128	4	4	32	100%	64 (100%)	8	0**	1
2.	B	128	512	64	64	128	100%	125 (50%)	8	1	1
3.	C	512	2048	250	250	300	100%	300 (30%)	8	2	1
4.	D	2 K	5 K	1 K	1 K	500	100%	500 (20%)	8	2	1
5.	E	5 K	15 K	2500	2500	1500	100%	1500 (20%)	8	3	2

* IP-PABX processor shall be capable to handle traffic up to expandable (i.e. Max.) capacity of IP-PABX without addition of any control card, RAM or memory. Only Analogue/ SIP port cards shall be required to be added.

**Requirement of PRI interface may be ascertained by the purchaser as the same may be required while expanding the IP-PABX capacity.

Note: IP-PABX should be capable of supporting any combination of SIP extensions and Analogue extensions .Total of both types of extensions shall be considered as the total capacity

11.0 Following shall be indicated in the Type Approval certificate:

REMARKS:

1. (a) Type(s) of Extension(s) used for testing
   (For Example SIP Phone/SIP Hard Phone/ Analogue phone)
   (b) SIP Terminal supported (IPv6and/or IPv4 should be indicated)
2. Types of interface supported e.g. Ethernet, PRI etc.
   For Ethernet interface IPv6 and/or IPv4 should be indicated
3. Types of Physical Ethernet interface
   (Ethernet type 10/100/1000 Map/twisted pair/coaxial/fiber etc.)
4. Additional codec supported, if any
3. Software required to read SDRs – Inbuilt/supplied separately
4. This TAC does not cover the functions, features, performance, capacity etc. of the extension equipments.
5. Approval of IP-PABX against this GR, does not entitle the user to connect the equipment to the network of Internet Service Provider's (ISP).

ABBREVIATIONS

AES	Advanced Encryption System
AOC–D	Advise of charge during the call
AOC–E	Advise of charge at the end of call
AMR	Adaptive Multi-Rate
AMR-HR	Adaptive Multi-Rate- Half Rate
CCBS	Call Completion on Busy Subscriber
CD	Call Deflection
CH	Call Hold
CLI	Calling Line Identification
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
CN	Comfort Noise
COTS	Commercial Type of Server
CPE	Customer Premise Equipment
CSCF	Call Session Control Function
CSH	Called Subscriber Held
CT	Call Transfer
CW	Call Waiting
Db	Decibels
dam	The absolute power level in Decibels
DDDS	Dynamic Delegation Discovery System
DEL	Direct Exchange Line
DID	Direct Inward Dialling
DISA	Direct Inward System Access
DoT	Department of Telecommunications
Does	Denial of Service
DTMF	Dual Tone Multi-Frequency
ECT	Explicit Call Transfer

EFR	Enhanced Full Rate
EMC	Electro-Magnetic Compatibility
EVRC	Enhanced Variable Rate CODEC
EVS	Enhanced Voice Service
FTP	File Transfer Protocol
GR	Generic Requirements
HDB3	High density bipolar of order 3
HTTP	Hipper Text Transfer Protocol
Hz	Hertz
I/C	Incoming
IEC	International Electro technical Commission
IEEE	Institute of Electrical and Electronics Engineers
IETF	The Internet Engineering Task Force
IDP	Inter Digital Pause
IM	Instant messaging
IP	Internet Protocol
IPS	Impulses per Second
ISDN	Integrated Service Digital Network
ISM	Industrial, Scientific & Medical
ISP	Internet Service Provider’s
ITU	International Telecommunication Union
LAN	Local Area Network
LDAP	Light weight Directory Access Protocol
LCL	Longitudinal Conversion Loss
MGC	Media Gateway Controller
MSN	Multiple Subscriber Number
MWI	Message Waiting Indicator
NAPTR	Naming Authority Pointer
O/G	Outgoing
PABX	Private Automatic Branch Exchange

PC	Personal Computer
PCM	Pulse Code Modulation
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
RFC	Request for Comment
RTP	Real-Time Transport Protocol
RTSP	Real Time Streaming Protocol
SBC	Session Border Controller
SDP	Session Description Protocol
SDR	Subscriber Data Record
SIP	Session Initiation Protocol
SRTP	Secure Real-time Transport Protocol
TAC	Type Approval Certificate
TCP	Transmission Control Protocol
TDM	Time Division Multiplexing
TEC	Telecommunication Engineering Centre
TFTP	Trivial File Transfer Protocol
TP	Terminal Portability
TX	Transmitter
UDP	User Datagram Protocol
URI	Uniform Resource Identifier
URL	Uniform Resource Locator
VMS	Voice Mail Service
VPN	Virtual Private Network
VoIP	Voice over IP

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