



Exploration & Production

GENERAL SPECIFICATION

TELECOMMUNICATIONS

GS EP TEL 111

Design and installation of an IP Telephony System
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1. Scope

The document defines the minimum requirements for the supply of an IP Telephony system which is the IP PBX in ToIP architecture. The number of technical points specific to each project shall be listed in the particular specifications of the project.

The contracting parties mentioned in this General Specification are listed hereafter:

- **COMPANY** The party which signs the contracts and purchase orders
- **CONTRACTOR** The fabrication CONTRACTOR on the yard or the construction CONTRACTOR on the final site
- **VENDOR** The supplier of the equipment.

The verbal forms used to indicate levels of requirement are specified in [ISO/IEC Directives, Part 2](#) and [Characteristics of good requirements](#), INCOSE symposium 1996.

2. Reference documents

The table below lists references from international organizations such as the EC, ETSI, ITU and others from national organizations such as the Telecommunication Research Centre of France Telecom (France Télécom R&D).

As a rule, the PBX shall be approved by the competent authorities of the country concerned. There follows a non-exhaustive list of norms and standards by which the PBX shall comply. However, exceptions can be accepted according to the characteristics specific to each country.

Where national regulations exist, their provisions and those of the standards and codes to which they refer shall apply, supplementing or amending the provisions of this document.

If there are no national regulations covering all or part of the subject of this document, the reference documents shall be strictly applied, as supplemented by the provisions of this document.

Only the main reference documents are mentioned; the CONTRACTOR shall be responsible for complying with all secondary reference documents dealing with the subject of this document.

The reference documents listed below form an integral part of this General Specification. Unless otherwise stipulated, the applicable version of these documents, including relevant appendices and supplements, is the latest revision published at the EFFECTIVE DATE of the CONTRACT.

Standards

Reference	Title
IEEE 802.3af	Power Over Ethernet: IEEE Standard for Information technology - Telecommunications and information exchange between systems - Local and metropolitan area networks - Specific requirements
IETF RFC 3261 - RFC 3262 - RFC 3263 - RFC 3264 - RFC 3265	SIP: Session Initiation Protocol

Reference	Title
IETF RFC 2508 - RFC 3544	CRTP: Compressing IP/UDP/RTP Headers for Low-Speed Serial links CRTP: IP Header Compression over PPP
IETF RFC 3246	Expedited Forwarding Per Hop Behavior
IETF RFC 3545	Enhanced CRTP: Enhanced Compressed RTP for Links with High Delay, Packet Loss and Reordering
IETF RFC 3550	RTP: A Transport Protocol for Real-Time Applications RTCP: A Transport Protocol for Real-Time Applications
IETF RFC 3435 - RFC 3660	MGCP: Media Gateway Control Protocol
IETF RFC 3711	SRTP: The Secure Real-time Transport Protocol
ISO/IEC Directives, Part 2	Rules for the structure and drafting of International Standards
ITU-T Recommendation T.30	Procedures for document facsimile transmission in the general switched telephone network
ITU-T Recommendation H.32x	Guide for Recommendations of the H.323 System
ITU-T Recommendation T.37	Procedures for the transfer of facsimile data via store-and-forward on the Internet
ITU-T recommendation T.38	Procedures for real-time Group 3 facsimile communication over IP networks
ITU-T G.114	Guidance on one-way delay for Voice over IP
ITU-T G.131	Talker echo and its control
ITU-T G.711	Pulse code modulation for Voice frequencies
ITU-T G.729	Coding of speech at 8 kbit/s using conjugate-structure algebraic code excited linear prediction

Professional Documents

Reference	Title
Not applicable	

Regulations

Reference	Title
Not applicable	



Other documents

Reference	Title
Characteristics of good requirements SCCP	Requirements Working Group of the International Council on Systems Engineering (INCOSE symposium 1996) Cisco SCCP

Total General Specifications

The provisions of this document shall be applied in the light of the supplementary documents supplied by COMPANY, including:

- Country specificity appendices
- Job specification
- Data sheets
- Articles and conditions; list of tasks
- Quality assurance provisions.

Reference	Title
GS EP INS 135	Cyber Security requirements for design and supply of ICSS and packages systems
GS EP TEL 002	Telecom Shutdown Philosophy
GS EP TEL 110	Design and installation of a PBX
GS EP TEL 170	Design and installation of communication networks
GS EP TEL 180	Engineering and installation of telecommunication network pre-cabling
GS EP TEL 700	Telecommunications Engineering Services

3. Functional definition

IP telephony systems are considered to be any primary voice communication system, which uses an IP network as the underlying transport for signaling and media.

An IP telephony system that is replacing an existing legacy PBX system would fall into this category, and in this case the new IP Telephony system shall respect the existing architecture. In particular latency and throughput between central telephony system and end-users shall be taken into account.

The primary goal of this architecture is to supply highly reliable, available, and secure voice services.

IP telephony also creates an opportunity to use applications that were not possible or convenient with legacy systems. Today's complex IP PBX are digital and with many servers attached to perform such functionality as voice mail, call center, access to new communications devices like wireless devices, Personal Digital Assistant (PDAs), and other applications.

The number and capacity of the equipment depend essentially on the number and nature of the terminals to be connected (telephone or data, analogue or digital terminals), and their geographic distribution. The processing capacity of the IP Telephony servers shall be calculated to allow for a 30% expansion over the initial load capacity.

The IP Telephony solution shall guarantee service availability for at least 99,999% of the global telephony service.

4. COMPANY Sites segmentation

It exists a large number of sites within the COMPANY environment.

Each site has a different number of users and may have cascaded sites behind it.

Others criteria like availability, reliability security shall be taken into account (redundancy, fall back scenarios) when designing the solution.

The CONTRACTOR shall propose an architecture that shall answer to the needs of every site but also propose as much as possible, a shared infra-structure, in order to regroup the IP telephony servers and to centralize services like unified messaging.

4.1 Centralized infra structure

In case of a centralized infra-structure proposed by the CONTRACTOR, this solution shall consist of several IP telephony servers and the CONTRACTOR has to enable this group of servers to work together in a cluster mode as a single IP PBX system.

The cluster shall hold its own configuration database, and shall provide:

- A Single point of administration for users for all sites, local or remote, within the cluster
- Feature transparency
- Shared line appearances
- Extension mobility
- Unified dial plan.

IP Telephony servers shall dispose of high performance and high availability capacities with as a minimum; RAID1 disk, dual power Supplies, dual fans and Dual NIC cards features.

4.2 Remote site IP Telephony Infra Structure

In some case, it shall be necessary to deploy local IP telephony servers locally on a remote site (Floating Platform Storage Unit for example regarding its geographical localization) in order to achieve service availability for at least 99,999%.

Remote sites that shall be connected and integrated to this core IP Telephony solution shall meet the following requirement:

- It is important to ensure that the WAN link between the data centers and the employee locations is fully redundant with no single points of failure
- It shall support a maximum **one way delay time** of 150-200 ms to meet the [ITU-T G.114](#) and [ITU-T G.131](#) recommendations for VoIP transport as well as voice mail traffic



- Packet loss shall be less than 1%
- Jitter should be minimized.

4.3 Production site

Some critical aspects can exist for production site.

Indeed high availability of the telephony can be a high expectation; production sites may need to perform outgoing calls permanently, they may need also to reach emergency number (fire brigade, 911...) and locally on the site they may need to call the security number.

Hazardous area can be present for some sites.

In off-shore site ToIP solution should have interfaces with backup link based on INMARSAT/BGAN system.

Interfaces with Point To Multipoint networks shall be required.

As per **GS EP INS 135** and **GS EP TEL 170**, industrial network is isolated from the business data and carried over dedicated VPN tunnel with encryption and authentication.

In some specific cases - when the latency, and/or available bandwidth don't permit to connect a site to a centralized telephony system, or if the 99,999% of service availability can not be meet, a remote site shall work in an autonomous way and shall host its own telephony system. This site in this case shall be connected to COMPANY telephony network through a gatekeeper gateway.

For these reasons the CONTRACTOR shall propose an architecture that shall be fit all requirements of production sites and shall manage the work authorizations permit on site following the site regulations.

5. Technical specifications

5.1 IP Telephony Infrastructure

ToIP solution design depends mainly of the number of users on site and the business requirement sites.

Most of the COMPANY sites are considered as critical in terms of security and business.

In case of a migration the new IP telephony solution shall allow a transition phase from a legacy telephony system (PBX based) toward a ToIP solution without service interruption during working hours. This ToIP infra-structure shall allow to adopt a flexible migration plan, site by site or user group by user group and according to the rhythm of the COMPANY.

The IT Telephony system shall inter-work with existing PBX system and with already deployed ToIP solution within the COMPANY.

The ToIP infrastructure shall be implemented over a communication network designed as per **GS EP TEL 170**.

ToIP solution shall support **at least** the signalization standard protocols (SIP - [IETF RFC 3261](#), [H.323](#) - [ITU-T Recommendation H.323](#), MGCP - [IETF RFC 3660](#), [SCCP](#)). The COMPANY recommends the use of [H323](#) protocol. The ToIP solution shall support **at least** RTCP - [IETF RFC 3550](#), CRTP - [IETF RFC 2508](#) - [RFC 3544](#), RTCP - [IETF RFC 3550](#), SRTP -



IETF RFC 3711 and Enhanced CRTP - IETF RFC 3545 transmission protocol, and ITU-T G.711, ITU-T G.729 codec.

The ToIP infra-structure shall allow to realize IP trunking with Telecom Operators (H323 trunking, SIP (IETF RFC 3261) trunking) and to inter-operate with other systems (at least H323, MGCP - IETF RFC 3660, SIP - IETF RFC 3261, Q.SIG).

ToIP solution shall offer service transparency, high redundancy and high availability.

Architecture proposed shall take into consideration the requirement of the COMPANY to centralize and share a maximum of services.

5.2 Circuit and traffic routing

The IP Telephony solution shall be capable of providing at least the following features:

IP Telephony servers shall be configured for optimized routing of the calls in order to:

- First of all: Optimize the cost of outgoing calls (based on the various rate grids for the PSTN access provided to the gateways)
- Secondly: Make the best use of the WAN network's available capacities (bandwidth)
- Route the emergency numbers.

And some function shall be implemented also like:

- Routing according to the dialed number (via the configuration of calling prefixes and route lists)
- "As close as possible" routing (routing according to the calling extension's location)
- Alternate routing & according to the time.

The IP PBX Manager can manage an alternative WAN/PSTN routing for voice and video calls. For example, in case of the WAN's unavailability or saturation (Call Admission Control), the call may be routed via the PSTN.

The IP Telephony System shall be able to be adapted to existing private networks numbering plans and to the public networks of the local country.

A proposal shall be made for DISA (Direct Inward System Access) in order to allow remote workers connection

In the proposed IP Telephony solution, voice mail access shall be done through the IP link preferentially but also through the PSTN.

5.3 Facilities offered to users

The IP Telephony system shall provide the users with the main telephony features, but not limited to:

- Call hold
- Broker's call
- Calling/called number presentation (CLI) on the display
- Call transfer
- Three party conference