ABC Company

Telephone System Replacement Project

DESIGN SPECIFICATIONS
Work Example

SPECIFICATIONS

1. GENERAL

Technical specifications are segmented between MANDATORY and DESIRABLE capabilities of the proposed system. Contractors must demonstrate the ability to provide installation, maintenance and support of all capabilities of their proposed solution.

2. MANDATORY SPECIFICATIONS

MANDATORY specifications are those, which the **Purchaser** must acquire in the proposed system. **Purchaser** has segregated mandatory specifications in the proposal form. Contractors are expected to identify their capabilities to respond to reach requirement and to explain any response with a partial or inability to comply in their proposal responses that might justify **Purchaser's** consideration of an exception."

3. DESIREABLE SPECIFICATIONS

DESIRABLE specifications are those, which the **Purchaser** desires to consider for inclusion in the contract. **Purchaser** has segregated desirable specifications for various reasons including fiscal budgets. In some cases, the feature capability may be scheduled for a future upgrade, or there may be nominal need for the capability now, but it is anticipated that the need may arise during the system life cycle. Each Desirable specification is listed and described and referenced by number. A description of each Desirable specification is provided on the proposal response form for convenience. Contractors are expected to identify their capabilities to respond to reach requirement and to explain any partial or inability to comply in their proposal responses.

SYSTEM FEATURES AND APPLICATIONS SPECIFICATION

Base telephone SWITCHING/ROUTING system

Contractors are to complete this form by inserting an answer that stipulates, "YES for full compliance, NO for non-compliance and PARTIAL for near compliance where applicable. Clarification of each answer is invited and essential for PARTIAL answers. If the Contractor has an alternative solution that will satisfy the obvious intent of the **Purchaser**, a detailed explanation is invited. Use a separate page referencing the section and item number.

Mandatory Specifications – Telephone SWITCHING AND ROUTING System

1. Telephone System shall be the manufacturer's most recent release at time of installation. Included in initial installation shall be any additional releases of software/hardware that the Manufacturer issues within six months of acceptance.
2. The system must offer converged voice and data using the same underlying technology as the Internet (TCP/IP), with voice being transmitted over local area networks (LANs) and wide area networks (WANs). The solution must be comprised of control elements, media gateway elements, applications, and a wide portfolio of desktop IP peripherals for small, medium, and large sites, as well as integration to support legacy TDM (Time Division Multiplexing) applications.
3. Contractor's suggested solution must be able to function with a variety of different manufacturers' legacy routers, firewalls, and switches on the LAN/WAN in an open and non-proprietary manner.

4. The system shall provide G711 and G729 compression on call transmission within the LAN/WAN network applicable to VoIP functions
5. Static IP addresses must also be accommodated by the System
6. Transmission delay must not exceed 100 milliseconds for the system. Delay across the WAN should be assumed to be a maximum of 125 milliseconds
7. The system architecture must offer converged voice and data using the same underlying technology as the Internet (TCP/IP), with voice being transmitted over local area networks (LANs) and wide area networks (WANs)
8. The system must be comprised of control elements, media gateway elements, applications, and a wide portfolio of desktop IP peripherals for small, medium, and large sites, as well as integration to support legacy TDM PBX ancillary servers and voice mail
9. The system must include a switched IP Core that is highly scalable and delivers reliable, robust switching, routing and call control – fully leveraging IP – while providing access to the PSTN using TDM technology
10. The system must also include a switched IP Core that is highly scalable and delivers reliable, robust switching, routing and call control – fully leveraging TDM technology for serving fully featured digital telephone sets in environments not supporting IP phones
11. Integrate with industry-standard LDAP corporate databases, without modifying existing corporate data and optionally allowing updates to existing corporate data
12. Call control must be provided to interface with the LAN access point, the phones, and handheld wireless computers to provide in-building mobile personnel access to voice and data that is functionally equal to that available at the wired desktop
13. The system must provide the capability to support Wireless IP Phones, as follows:

13.1. The system must run on a 802.11b wireless LAN and support G.711 and G.729 Compression.

13.2. The system must be capable of being enabled by adding only access points, wireless phones and software licenses.

13.3. The system must afford integrated full featured call control comparable to wired phones

13.4. The wireless handsets must be ACD capable

13.5. Ability to “twin” wired phone with wireless handset (i.e. one extension number rings both phones, they share a mailbox, and voice E-mail notification lights both message waiting lamps.

14. The system should use industry-standard database technology for internal directory management
15. The system must support Q.SIG integration features
16. Switching must be provided via the enterprise’s IP network. for IP-device-to-IP-device communications, e.g., calls between Ethernet IP phones,
17. IP networking features must allow office-to-office voice traffic between multiple controllers to be routed over **Purchaser’s** LAN/WAN
18. The system should include a web-based interface available for IP telephone users to program their own telephone feature keys
19. Switching must be handled via a conventional circuit-switched (TDM) bus for TDM circuit communications, e.g., a call between an analog phone and the PSTN, or between the PSTN and a stand-alone voice mail application,
20. The architecture must package the traditional circuit-switched bus, call server, gateway/gatekeeper and real time applications resources within a 19” rack
21. The system must be able to support both digital and analog trunks for connection to the PSTN or for connecting multiple sites or systems together

22. The system must be capable of networking systems together using traditional circuit switched technology or the IP infrastructure
23. The system must be able to be used as a standalone application, e.g., telephone switching and routing system, Voice Mail System, or a mobility solution
24. The system must be capable of using an embedded DHCP server and an FTP server when required. There must be an option to disable this when the resources are available on the enterprise network
25. Hot Desking: The system must be able to provide a "Hot Desking" function, whereby a pool of phones may be shared by a number of users and a user has a specific Directory Number, but not a specific phone.
26. Hot Desking: When the user logs into a shared phone, the user's attributes are assigned to that phone. Attributes would include: directory number, class of service, call forwarding, voice mail and message waiting, feature keys, speed dial keys, and call restrictions
27. The station hardware must provide for basic convince phone to multi-button display phones suitable for a dispatch station.
28. System shall have deployed or scheduled deployment within 6 months of SIP based protocol phones.
29. Station Connectivity: Supported connections must include these options:

29.1. A 10/100/1000 BaseT Ethernet connection is required for the IP sets to connect through an Ethernet LAN to the system.

29.2. Line power to an IP phone via the Cat 5e cabling, either supported by ABC Company active electronics or active electronic upgrades (switches, power hubs, etc.) provide by the Contractor.

29.3. On-premises analog station ports (24V per port) are required for industry standard DTMF analog telephone sets. The external loop resistance on these stations must be 600 ohms or less, and the loop length must be 3,000 ft. (1000M) or greater on 26-gauge wire.

29.4. Off-premises analog lines (48V per port) for industry standard sets or devices where the external loop resistance exceeds 600 ohms or where lightning surge protection is required. The maximum resistance on these must be 1800 ohms or less, and the loop length must be 12,000 ft. (4000 m) or greater on 26-gauge wire.

29.5. Digital Interface ports must provide the interface for system digital telephones and consoles.

30. Network Connectivity: Supported connections must include these options:

30.1. Analog Trunks

30.2. Loop Start/Ground Start (via peripheral);

30.3. E & M (via peripheral);

30.4. Direct Inward Dial (DID);

30.5. Loop/Tie trunks.

30.6. Digital Trunks

30.7. T1

30.8. ISDN Connectivity: The system must support multiple Integrated Services Digital Networking (ISDN) protocols and provide ISDN connectivity. The system must connect to the ISDN public network and data devices (such as routers, video conferencing equipment, and servers) by using PRI or BRI. The system must be able to take advantage of the following features to capture and control costs, analyze peak periods, and fine tune network resources accordingly for both voice and data calls.

30.8.1.1. Primary Rate Access (PRI)

30.8.1.2. Basic Rate Access (BRI).

30.9. VoIP Ethernet

30.10. Point-to-point T1;

31. Digital Interface ports must provide the interface for TDM system digital telephones and consoles.
32. ARS/LCR (Automatic Route Selection/Least Cost Routing);
33. CDR (Call Detail Recording); The capability to connect a call accounting system via a CDR (Call Detail Recording) port and the provision of internal, local and long distance call detail data.
34. The telephone switching/routing system affords comprehensive Toll Control as an integral part of the Call Control. It should allow restriction of user access to trunk routes and/or specific external directory numbers. It should also allow Class of Restriction (COR) and Class of Service (COS) features that could substantially reduce the risk of toll fraud. The toll restriction allows the ability to assign the dialed number with an authorization code.
35. Networking Feature Transparency when using switched (call by call) ISDN.
36. IP Networking Connectivity
37. IP networking features must allow branch-to-branch voice traffic between multiple controllers to be routed over the organization's LAN/WAN infrastructure
38. The system offers an optional board/meeting/office audio conference unit
39. The system must provide fully featured call control services using an industry standard real time operating system. In this context, fully featured means all of the features and functionality traditionally associated with a TDM voice switching/routing system such as multiple levels of call forwarding, message waiting, advisory messages, conference calling, account codes call barring, least cost routing, night service, and so on.
40. TAPI™ Microsoft's TAPI (Telephony Application Programming Interface) must be supported for desktop applications or server applications
41. Microsoft's Wav must be integrated and supported in a desktop and server format for application development requiring audio
42. The system must offer a proprietary Telephony Applications Interface that allows 3rd party developed CTI (Computer Telephony Integration) applications to interface with the telephone switching and routing systems all Control elements. A future deployment of an Interactive Voice Response system application may apply to this functionality.
43. Reliability

43.1. The switching/routing core must be non-blocking in wire capacity configuration.

43.2. Voice traffic must have priority over data and other non-real-time communications traffic on a VoIP environment.

43.3. The system must support QoS (Quality of Service) and the Standard 802.1p for voice prioritization.

44. Fault Resiliency

44.1. The system must provide the ability to configure controllers in a primary controller / secondary controller configuration. The secondary controller must be able to provide call control services to all phones on the primary controller in the event that the primary controller fails. Phones must be able to auto-register with the secondary controller in the event that the primary controller fails. When the primary controller is back on-line, the phones must automatically re-register with the primary controller. There must be no loss of service when control is switched from the primary to the secondary controller.

44.2. The secondary controller must be configurable in a geographically dispersed (i.e. across the WAN) location to provide an even greater degree of protection against disruptions at the primary site (i.e. flooding, etc.).

44.3. A controller must be able to function as the primary controller for a group of phones and act as a secondary controller for another group of phones.

45. Maintenance and printer ports, alarm indications, and SNMP (Simple Network Management Protocol) must also be embedded in the system
46. The system must provide for the configuration of system attributes such as voice mail, trunking, DHCP and ACD
47. The system must provide trunk Diagnostics; for both digital and analog services
48. Authorized access to the system tools must provide protection for various administration commands from unauthorized users. Each user must have a log-in name, password, extension number, and language preference.
49. Maintenance logs must be retrievable
50. The Telephone System must support MAT (Maintenance and Administration Terminal) software that will facilitate administration of the systems through the ABC Company LAN/WAN. Additionally, ABC Company will be supplied with MAT software that can be installed on a **Purchaser** supplied PC for local direct access to any Contractor supplied switching/routing node.
51. The system administrator must have the ability to perform diagnostic tests on system resources
52. Remote System Maintenance Access. The System shall be equipped with remote access via dial-up modem or LAN/WAN MAT terminal for remote maintenance.
53. Alarms must be raised at certain thresholds of failure or error. The attendant console must be signaled in each case of an alarm
54. There must be diagnostic tools available to troubleshoot problems, such as testing of voice quality on phones
55. The remote network administration must have the ability to view alarms, and obtain an alarms history.
56. The system must enable the system administrator to manage multiple nodes simultaneously, across multiple sites
57. The system must be programmable to provide scheduled MACs
58. The system must provide scheduled backup of management and node data
59. The ability to program the outgoing call line access number as 7 or 8 instead of 9.
60. The system must have the capability of integration to the existing overhead paging system via station or trunk ports so that various paging scenarios can be activated via any telephone station.
61. E 9-1-1 and 9-1-1 calls from any telephone within the proposed systems must be capable to identify the physical address of the calling station to the PSAP with the use of <u>optional</u> CAMA trunks or PRI with the necessary database maintenance, for example through PSALI services.
62. Provision for 911/E911 Service: The system must be optionally capable of successfully and accurately passing extension number and location details to the PSAP. Describe the optional configuration solution currently offered.

62.1. Connectivity and transmit to PSAP options:

62.1.1. SDN ECS over PRI

62.1.2. ELIN (CESID) over DTMF via a third-party converter

62.2. Telephone Directory Capability for Location Identification

62.3. Embedded On-Site Notification – 911 calls must to a minimum raise an alarm at the Dispatch station or defined display sets attached to the PBX for on-site staff to assist emergency response personnel.

62.4. E911 Support for other emergency numbers: Emergency calls are not limited to 911. The solution must support any digit string that can be designated as emergency (e.g. 3333 is a violent patient notification in healthcare), and sent to the attendant console or defined display sets.

62.5. E911 Logging and Reporting: All 911 calls must be logged and be exportable for reporting purposes. Describe.

62.6. Support for E911 Dispatching options - The capability for on-site and off-site. 911/E911/Emergency notification via pagers, cells, wireless, cordless, wired, etc. and the announce mechanism.

63. The system must deliver head office features for both voice and data at the teleworkers' locations, via an IP connection to the remote IP Phone. The PC must connect to the company LAN via the IP phone. The remote IP Phone must be easily configured at the remote site. Access to the corporate LAN must be secure
64. The Telephone System shall have Computer Telephony Integration (CTI) integration capability
65. The ability to adjust the audio volume on telephone sets and attendant console.
66. A methodology for managing calls in the event of a power failure exceeding the battery backup capacity.
67. Toll fraud protection plan and associate features to prevent unauthorized people from breaking into the system and dialing out to anywhere.
68. The optional capability for a station user to initiate a six or more party conference including both internal and external callers with optional password protected access.

Attendant PC Workstation Optional Capability: Support for Centralized Attendant Services (CAS) and Network Attendant Services (NAS). Displays call processing status, directory and related information windows for call control and monitoring.

Incoming Caller ID: Visual indication of Caller ID with the capability to forward the original Caller ID with a transferred call and again on a subsequent transfer.

Outgoing Caller ID:The system transmits the DID number identification of the outgoing caller's station, with the option of blocking on a station-by-station basis.

Direct Emergency Dial Numbers: Ability to dial a set of emergency numbers like 911 without having to dial an access code, and guarantee of outside access availability.

Distinctive Inside / Outside Ringing: Different ringing tones for inside versus outside calls.

Do Not Disturb: Calls will not ring and transfer immediately to voice mail or coverage.

Extensions List: Visual display of all extensions with names/location and associated numbers of the phone system.

Group Call Pick up: Pick up an undetermined ringing station from another station.

Group Extensions and Ringing: Groups of stations can have the same extension number.

Individual Call Pick up: Pick up a specific ringing station from another station.

Message Waiting Display on Telephone Set Indication of voice mail messages is unread.

Multiple Call Handling: Ability for a single station to handle 10 or more simultaneous calls.

Multiple Extension Assignments per Station: Each station can have more than one extension number assigned.

Multiple Redial: Redial the last 10 numbers called.

Mute: Mute mouthpiece or microphone of phone.

Personal Telephone Directory: Ability for each station to have a personalized list of frequently called numbers.

Recent Callers List and Call back: List of 20 or more recent callers and the ability to call them back. Indication if call was answered is preferable.

Redial: Redial the last number called.

Telephone Call Control Integration: Ability to integrate call control functions into other PC applications or Web browsers.

Three or more Party Conferencing: Can initiate a three or more party conference from the station.

Transfer Indication: Indication of where incoming call has been transferred.

Visual Indication of Parked Calls and Time on Park Station to have indication of park zones and the time the call has been placed in the park zone.

69. The capability to selectively program a station to ring more than the standard ring count before routing to voice mail, with the flexibility to set the ring count by day or time of day.
70. The capability to deploy portable IP phones, with a cell phone like form factor, that can function as telephone stations that access the voice switching/routing platform via existing wireless data access points.

Desired Capabilities – Telephone SWITCHING AND ROUTING System
71. A "one number" capability for users, based on a class of service, to have the phone system ring both their cell phone and their desk phone simultaneously. Upon the user accepting the call at either appliance, then the system would drop the other attempted route call delivery.
72. A visual notification of the second call waiting or to be transferred while a station user is on a call.
73. Variable Ring Sound. Ability to distinguish between the sounds that the ring makes. Ability to provide a range of sounds that the ring makes
74. The capability for the system to capture caller ID information and store it in a database for non-real time manipulation offline.
75. The ability to initiate a page to another phone station and override the user status to intercom page that user over their phone's speakerphone.
76. The system should provide a web-based interface that enables the administrator to make changes to user information (e.g. Add or delete Hunt Group members).and provide comprehensive Move, Add and Change (MAC) functionality, including templates and batch MACs
77. The system should have the capability to page a system technician upon critical alarms events
78. The system should provide an On-Line Help facility
79. The ability to integrate a wireless headset with a single interface for both radio and phone access.
80. A voice activated dialing capability similar to how cellular companies offer voice recognition features for dialing names in the phone directory.
81. The ability to completely mute the ringer of the telephone set.
82. A TTY option at the phone set.

Voice Messaging System Specifications

Mandatory Specifications – Voice Messaging System
83. Telephone System must integrate with a centralized **Voice** Messaging **System (VMS)**. *The* Voice *Messaging system shall support the following minimum requirements:*
84. LAN/WAN based access for system administration. The VMS shall be equipped with Maintenance and Administration software that will facilitate administration of the systems through the a WAN. Additionally, software is supplied that can be installed on a customer supplied PC for direct access to the Voice Mail system from any port on the applicable LAN/WAN.
85. Optional capability for Unified (Universal) Messaging, VPIM compliant.
85.1. Integrated Fax Mailbox option.
85.2. Multi-lingual mailbox option.
86. The VMS will automatically set a visual message waiting indication on the subscriber's telephone set (at the host system site and remote switch sites) or if the telephone set is not equipped with a visual indicator, then stutter dial tone must be provided to indicate to the user that a message is waiting.
87. Station users may directly forward callers to voice mailboxes when a telephone extension is

	Mandatory Specifications – Voice Messaging System
	busy or not answered.
55.	Callers forwarded to the VMS will be able to press a DTMF key to escape to an operator or to the automated attendant for accessing another extension.
88.	Automatic Overnight Maintenance.
89.	Backup and restore capabilities.
90.	Toll Fraud control capabilities enabling the system administrator to manage toll fraud risks.
91.	Capacity to serve at least 350 users, expandable to 500 without limitation on the number of messages that may be stored in a mailbox, with the capability of the system administrator to assign class of service limits to each mailbox.
92.	The system must be capable of providing Unified Messaging using one of the following methods:
	92.1. The unified messaging application must offer enhanced features and functionality to the employee who is travelling and is away from the office most of the time, e.g., calendar, contact list, meeting scheduler, tasks, audio E-mail, visual voice E-mail access
	92.2. The unified messaging application must offer the option of MSS (Message Synchronization Service) which provides redundancy by duplicating messages in various storage locations to track all changes in message status, and provides full synchronization of all message types
	92.3. The system must accommodate deployment in any IMAP4-compliant E-mail environment, such as Exchange 2000 or Lotus Notes
	92.4. User access and management of E-mail, voice mail and fax mail via phone, E-mail client or web interface;
	92.5. User administration of their own selectable features such as paging, personal automated attendant, message forwarding, etc.
	92.6. The system must provide an optional feature of text to Speech E-mail reading for remote access
	92.7. The system must optionally support networking to other unified messaging or voice mail systems via VPIM
93.	The system offers a speech enabled automated attendant directory
94.	The ability to archive and store some voice messages indefinitely.
95.	The ability to program multiple message waiting notification appearances on a single telephone station. This requirement relates to a common area where multiple phone users have access to a common phone set but need to determine individual visual message waiting notifications.

	Desirable Specifications – Voice Messaging System
96.	Text-to-speech – the capability of the system to speak textual content such as E-mail messages in an Unified Messaging scenario.
97.	The ability to program a “question and answer” mailbox that can be programmed to ask a series of questions and record the respondent’s answer as a single message with recorded delimiters for each question answered.
98.	The ability to activate the recording of a call as a voice message.
99.	The ability to use voice commands (voice recognition) to control system functions in lieu of DTMF (touch tone) commands.

Automated Attendant System Specifications

Mandatory Specifications – Automated Attendant
100. Supports multiple automated attendants, accessible manually by dialing a mailbox number or automatically through Telephone System call routing routines.
101. Multiple directory groups, one for each department with Automated Attendant capabilities.
102. The ability for callers to reach any extension by dialing the extension or accessing an automated attendant directory feature to dial by name. The caller would dial the first few digits of the called party's first or last name as programmed by the system administrator.
103. Automatic Time of Day, weekend, holiday scheduling of after hours call processing with separate greeting options for time scheduled call handling which will be automatically selected.
104. DID addressable, multiple alternate automated attendant scenarios. For example a department may create its own, unique and separate automated attendant scenarios by programming an Automated Attendant mailbox with its own menu choices such as transfer to another extension, mailbox or even a speed dial code.
105. Call screening option, assignable as a class of service feature. Call screening enables presentation of the calling party ID and allows the called party the option of accepting or rejecting the call.
106. Single key or multi-key options. Programmable to enable callers to enter an extension or limit callers to only a single key choice from a menu.
107. A capability for the system administrator to establish informational mailboxes with single digit exit options to other mailboxes or pre-designated extensions and/or the ability for the caller to access an automated attendant directory and dial a specific extension.

Desirable Specifications – Automated Attendant
108. Off line database administration. (The ability of the system to import and export a directory database for off-line creation update and administration. 109. The ability to allow a caller to select an option at the automated attendant that would put the caller into a call pickup queue and then page the intended recipient (or department). Then take the call back if no response.

ACD (Automatic Call Distributor) Specifications

Mandatory Specifications - ACD
110. The ACD shall have the capability of programming multiple queues such as for alternate languages, special ADA situations or other specialized services.
111. The ACD must have the capability of pre-programming for high call volume events or disaster recovery situations to process queues on a priority basis. For example the calls to a "Snow Day" queue can be processed and completed faster than all other types. Accordingly, that queue might be assigned priority during unusually high call volume events.
112. The ACD will have the ability for ACD supervisors, managers, etc. to observe quality of customer service between the agent and callers from a telephone set without side jacking or creating awareness to the caller.
113. Networked ACD must make it possible to integrate the operations of two or more IP based voice switching/routing systems

Mandatory Specifications - ACD
114. Visual queuing for ACD agents. The ability to visually display calls in queue for multiple (minimum of 3 queues) queues. This application is intended to allow a dispatcher the ability to selectively choose specific incoming calls based on current emergency conditions. The Dispatcher will have up to 3 queues of calls (customers, emergency agencies and ABC Company staff) at any time. Normally, calls will be answered in the order received; however, during emergency situations, the Dispatcher needs the ability to visual review all callers in the various queues and selectively answer a specific call.
115. Intelligent Recorded Announcement Devices: The ACD supervisor must be able to customize Recorded Announcement Devices (RADs) messages that callers will hear based on the time, date, and/or number of callers in the queue. The supervisor must be able to manage the announcements via a browser-based interface with .WAV file recordings.
116. The ACD must have functional capabilities including:
116.1. Skills based routing
116.2. Order of arrival queuing
116.3. Abandoned call removal
116.4. Overflow to secondary groups
116.5. Call transfer
116.6. Priority Queuing
116.7. Remote diagnostics
116.8. Overflow routing of incoming calls
116.9. Announcements on hold: Note: the RAD(s) (Record and Announcement Device) must be functional at each ACD group physical location in the event of a disruption of service. For example, if a trunk group (A or B) and the private voice/date line between OTC and the main transit facility is disrupted, RAD support is continued.
116.10. Delay announcement. Note: the RAD(s) (Record and Announcement Device) must be functional at each ACD group physical location in the event of a disruption of service. For example, if a trunk group (A or B) and the private voice/date line between OTC and the main transit facility is disrupted, RAD support is continued.
116.11. Each ACD supervisor must have a unique login ID
116.12. Each ACD agent must have a unique login ID
116.13. The Contractor must have a maintenance unique login password
116.14. The system administrator must have a unique login password
116.15. After call work button
116.16. Incoming call ID
116.17. Assistance button/dial code
116.18. Emergency indicator
116.19. Call wrap up
116.20. Operator headsets (Starkey compatible)
116.21. Alert with incoming call
116.22. Distinctive Ringing
116.23. Work state button
116.24. Call waiting indicator
116.25. Mute (cough) button
116.26. Hold button
116.27. Individual logon identification

	Mandatory Specifications - ACD
	116.28. Ability to prioritize Trunks
	116.29. Ability to research historical data
	116.30. Ability to move agent queue assignments dynamically
	116.31. Ability to set queue priorities
	116.32. Ability to conference with agent and caller
117.	ACD Reporting Capabilities: The ACD must have the capability of creating statistical records (with date and time stamp) on a periodic basis for management reports without taking the system system-down. At a minimum, the statistics detailed must be programmable at any interval. Reports give supervisors snapshots of the ACD's performance and status. This allows them to react appropriately to evolving conditions. Abandoned calls, for example, can be monitored to determine the waiting-for-service tolerance of callers compared to the number of calls in queue. Additionally, agent productivity can be compared at a glance to determine who may need help in speeding up after call work. Each ACD will require real-time reporting that shows all activity. At minimum the real-time reporting should include:
118.	Agent Reports: Should include a list of all active agents during current interval (i.e.: 30 minutes), the current activities of all agents (what queues their assigned) and the status of each active agent (available, on a call, on break, etc.). Ideally real-time agent reports would include the following: agent name, queue assignment, current status, how long in that status and total calls taken this interval
119.	Queue Reports: Should include a list of all activities for a specific queue during the current interval (i.e.: 30 minutes), the current activities of the queue including how many calls are currently waiting in the queue, how many calls have been answered and how many calls have abandoned in the current interval. The real-time queue report would ideally also show how long the oldest call waiting has been holding in queue.
120.	Historical Reporting: Historical reports display, report and summarize the past performance of any measured subset of the ACD. Historical reports display past data for various agent and queue activities, such as number of ACD calls, abandoned calls, average talk time and average speed of answer.
121.	Queue Reports: The Queue report gives detailed information regarding the performance of each queue. Historical queue reports include the following per the interval requested: total number of calls queued, total answered, average speed of answer, average talk time, total time on active calls and maximum wait time during that interval.
122.	Extension Report: The extension report shows the amount of time that each agent spent talking on his or her personal phone extension. This information serves useful for supervisors and managers addressing time management efficiency in the ACD group.
123.	Reason Codes Report: Reason Codes allow contact center supervisors and managers to determine how agents are spending non-call processing time. This report details the amount of time spent associated with each "Unavailable" or "Not Ready" activity, for example, training or break times.
124.	Announcement Play Reports: The Announcement Play report displays the total number "plays" for each announcement for the interval requested.
125.	RONA Reporting: RONA (Redirect on No Answer) reporting allows contact center supervisors and managers to determine how often a call was prompted to an available agent where the agent did not answer the call.

Mandatory Specifications - ACD
126. **Business Continuation:** The ACD must have the capability of supporting business continuation under various disruptions of service conditions. If the PSTN connectivity is lost at the OTC (Olympia Transit Center), and the private line (T1 between sites) is lost, or if during special projects or snow days, the customer service agents at OTC would be able to log on at available workstations at the main transit facility and continue customer service. Conversely, if the PSTN connection at the main transit facility and the private line is lost, the Dial-A-Lift agents would be able to log on at available workstations at the OTC and continue service. Recognition is given to the reduction in call handling traffic capacity in such events. Note: If this functionality involves additional costs, identify such additional costs in **Error! Reference source not found.** - **Error! Reference source not found.**.

Desirable Specifications – ACD
127. The ability of the system to detect the number of callers in queue and ACD call processing statistics, and then announce to the next caller the estimated wait time until answer.
128. The ACD should have the capability of a digital voice and data recording system that will allow selective or full-time recording of each caller transaction for training, or other purposes. The recording capability coordinates monitoring, logging, evaluation, archiving and retrieval functions of agent activity. Plus, it should enable integrated monitoring, evaluation, reporting and coaching tools for contact center supervisors. It should also have the capability for inbound and outbound voice applications for playback, live access, voice response, conferencing and notification.

CONTRACTOR

Mandatory Specifications – Contractor Support
129. The Contractor must offer basic system administration training. The level of training should include items like station relocation, class of service changes, changes to speed call list, pick up group changes, hunt group changes, Voice Mail and any other components provided by the Contractor.
130. An option for ABC Company to purchase and enroll two Information Systems technicians in manufacturer provided technical training for system maintenance and database programming
131. The Contractor must be able to include the preparation of a training room, if required (i.e. extra wiring, set-up for training provided at ABC Company.
132. The Contractor must provide classroom training for all station users, system administrators and hands-on training for attendants.
133. The Contractor must provide specific hands-on classroom training for up to 20 ACD agents.
134. Availability of local, reliable, dependable, and ongoing maintenance support by the system Contractor.
135. A flexible set of alternatives for user reference and tutorials. The Contractor must provide reference guides on how to use the phone system. In addition to initial hands-on training and user guide documentation, access to help features via the telephone display and optionally access to an Internet or Intranet based interface for instructions on how to use telephone set, ACD, voice messaging and automated attendant features.
136. The Contractor must determine and specify what product or skill certifications (MSCE or

Mandatory Specifications – Contractor Support
other) will be required of ABC Company so it may support the Contractor’s product.
137. The Contractor will be required to conduct a station review of all intended telephone station or peripheral device interfaces and determine the infrastructure readiness for the type of telephone device to be connected. If the cabling or active electronics infrastructure does not support an IP phone, then the Contractor must provide a suitable alternative such as a proprietary phone or POTS device.

CONFIGURATION SPECIFICATION

Telephone Switching/routing System						
	Equipped [1]		**Wired [2)]**		**Maximum [3]**	
	Main [4]	OTC[5]	Main	OTC	Main	OTC
DS1 – (PRI) 23 Trunks[6]	2	2	4	4	8	8
DID numbers	300	50				
Analog Trunks Loop/Ground Start	4	4	8	8	8	8
Analog Station Ports OPX Circuits	0	4	4	4	8	8
Analog Station Ports	24	8	36	16	48	24
Voice Recording and Logging System Ports	8	8	16	16	16	16
VoIP Or Digital Phone Station Ports	110	18	144	24	196	48
Voice Mail System Ports[7]	12	N/A	24	N/A	24	N/A
Paging Interface Ports	4	2	4	4	8	8
Power Failure Transfer Stations	1	1	2	2	4	4
Attendant Consoles	1	N/A	2	1	2	2
ACD Groups	1	1	16	16	24	24
Intercept Announcer	1	1	1	1	2	2
CSU	2	2				

Voice Mail System			
	Equipped (11)	**Wired (12)**	**Maximum (13)**
Ports	12	24	24
Hours storage	500	500	1000

[1] Active capacity equipped upon initial installation

[2] Wired capacity enabling upgrades by adding blades and/or software to increase capacity

[3] Maximum physical hardware/software expected over the life of the system

[4] Main -= Main Transit Facility

[5] OTC = Olympia Transit Center

[6] Includes SCAN access trunks

[7] Assumes Voice Mail/Automated Attendant system shared and housed at Main Transit Facility

ACD									
	Equipped			Wired			Maximum	
Agent software licenses	12	12		48	48		72	72
LAN interface via Ethernet port	1	1		1	1		1	1
ACD Reporting Software	1	1		1	1		1	1
ACD Interactive Management	1	1		1	1		1	1
ACD Scheduling Application								