VoiceCon Request for Proposal for an IP Telephony System

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Preface

The following RFP document was exclusively designed and developed by TEQConsult for the VoiceCon® Spring 2007 Conference.

The RFP is intended to solicit product information and pricing data about IP Telephony systems during the Fall 2006 time period. The RFP was written for a large multi-facility enterprise configuration with IP voice terminals as the primary station user interface to the system. TEQConsult Group recognizes that every business and institution has unique communications needs and resources, but the much of the material included herein can be used by VoiceCon workshop attendees regardless of their unique system size and application requirements.

VoiceCon workshop attendees may use this RFP as a template for customizing their own RFP with the proviso that proper accreditation to TEQConsult Group will be included in the document.

TEQConsult Group would like to thank Fred Knight, VoiceCon GM and the publisher of Business Communications Review, for his review and editing of this document, and to Unimax Systems Corporation for its contributions to the systems management section of the RFP.
Siemens Executive Overview

The Siemens team appreciates the opportunity to respond to this request for an enterprise telephony platform, and is confident that our solution will provide a flexible and cost-effective means for VoiceCon to deliver telephony services and advanced communications applications in an on-demand model. Siemens is proud and excited to deliver the following solution that will meet and exceed VoiceCon’s vision for IP Telephony.

Recommended Solution:
Siemens is proposing the HiPath 8000, a real-time, IP-system based upon SIP for the enterprise providing carrier class reliability, survivability, and scalability. The HiPath 8000 scales to 100,000 users per instance and to an unlimited number of users per network (depending upon configuration). The HiPath 8000 employs an IT/data center centralized model that supports distributed environments. Highlights of the enclosed proposal include, a dual server HiPath 8000 with the servers geographically separated between the HQ and RO to provide for full HiPath 8000 functionality even if the HQ WAN link should be lost. Fully survivable gateways at all locations to provide for telephone functionality should the entire WAN/LAN network fail. Siemens OpenStage phones which provide the latest technology at the desk top. The Siemens’ platform delivers distinct advantages in meeting VoiceCon’s requirements, far exceeding LAN-based and IP PBX-based approaches by providing:

- **A highly scaleable, carrier grade enterprise telephony platform that extends telephony services seamlessly and globally.** Siemens’ simplified architecture results in substantial reduction in TCO and unsurpassed flexibility in deploying global telephony services. For example, robust route selection and hop-off capability enable global access to the most cost effective public network resources. Thus, Intel’s telecommunication costs will be reduced not only for internal calls completed on the IP network, but also for local, regional, and international toll calls. As well, the HiPath 8000 leverages current infrastructure investments due to industry standard interfaces, standards and languages. Migration from existing Real Time IP Systems can take place in a step by step manner. Existing Real Time IP Systems can be used as media gateways with HiPath 8000. Even existing third party PBX systems can be included when supporting the Q.SIG industry standard. This makes the openness of HiPath 8000 especially cost-effective.

The following table, based on a study conducted by Forrester Consulting for Siemens Communications, shows Key ROI Drivers and Business Value provided by Siemens’ HiPath 8000 solution.
• **Truly global deployment and support resources.** Siemens Communications is unmatched in its ability to *directly* fulfill the geographic requirements VoiceCon has specified in the RFP and is fully competent to directly complete such a massive project as well as to train and provide “on the job” experience to VoiceCon resources globally. The HiPath 8000 is a software application that resides in and will be managed like any other traditional data application to provide a real-time communication service delivery platform. This enables Siemens to offer a wide range of services to support the deployment, maintenance, and management of Intel’s solution directly. Our services offerings range from basic remote software maintenance of the HiPath 8000 system software, through multi-vendor maintenance and remote management, to complete management of voice premise and end-user services.

• **Centralized management creating a common user experience.** A single system meets the specific needs of Intel’s geographies and divisions. Competitors’ approaches typically network disparate systems that must be administered with a variety of management tools. Competing approaches typically network disparate systems that must be administered with a variety of management tools. This architecture not only increases costs, but also creates an environment that is inherently difficult to configure, provision, secure, and troubleshoot. Siemens’ services-oriented approach extends centralized telephony functions to a wide variety of applications, and facilitates consistent access from the interfaces and devices that best fit the user’s current context and location. The result is a global unified communication experience for users across mobile and wired networks, devices and applications. It provides VoiceCon employees greater access to the people and information they
need to innovate, respond to customers, and execute business processes effectively.

- **Carrier-grade resiliency.** Siemens has successfully delivered the SIP platform to large scale global carriers for many years. The HiPath 8000 has two nodes (i.e. two servers) that are inter-connected via multiple IP paths over separate physical connections. Software specifically tailored for carrier grade real time applications ensures that the nodes are in active-active running order. Failure of a node causes no interruption or feature loss. In contrast, a LAN-based IP Telephony competitor uses a database replication system and must use other proprietary mechanisms to share state around the 'cluster' (the number of servers that actually forms the implementation). The HiPath 8000 is based on Linux, not Windows. There is no system administrative overhead with regard to patch management associated with Windows. Consequently, the HiPath 8000 meets the most stringent uptime criteria of a carrier grade platform—criteria that our enterprise competitors cannot match.

- **SIP-Based Architecture.** Siemens supports RFC3261 compliant handsets today, unlike our competition. The HiPath 8000 supports any 3rd party SIP-compliant phone or soft client. We offer a variety of Siemens SIP telephones and our soft client, the optiClient. Of course, we can certainly operate in a mixed environment with both our devices and other SIP devices. VoiceCon would therefore able to deploy the right mix of cost-effective and advanced hard phones, soft phones, and mobile phones to meet its needs without regard to the limitations or cost points of any one vendor. Because phones are one of the most costly per user components of a telephony solution, this flexibility provides VoiceCon with a significant opportunity to drive down acquisition and ongoing support costs.

- **Multi-tenant Capability.** The HiPath 8000 can be partitioned to reflect different legal organizations within a company. Business units can be reflected as separate entities within a single HiPath 8000. Our competition does not support this capability natively. They require external 3rd party applications, which create another potential area for performance issues surrounding the interface, as well as potentially higher costs.

Siemens HiPath 8000 can support VoiceCon in both a wired and wireless environment, providing a common user experience regardless of connectivity. Regardless of whether users are connected via a public or enterprise network, or wired for wireless, their telephony experience will be transparent from one connection to the next.

**Summary:**
Siemens welcomes the opportunity to collaborate with VoiceCon on its enterprise telephony transformation. The depth of our innovation across enterprise, service provider, wired and wireless networks ensures delivery of a compelling solution. We look forward to this opportunity to share our vision and fully recognize the significance of the global project you are undertaking.

Thank you for your consideration-- we look forward to expanding our mutually successful working relationship in years to come.

VoiceCon IP Telephony System Request For Proposal

General Guidelines for Proposals

1. Please read though the entire RFP before beginning to work on your response.
2. Configure and price your system design to satisfy all stated RFP requirements, including any and all system hardware and software elements necessary to satisfy a requirement. Vendors that underconfigure their system design to reduce its price proposal will be penalized.
3. All products and solutions proposed for this RFP must be formally announced as of January 15, 2007 (prior to VoiceCon Spring 2007).
4. Do NOT provide material or information unrelated or not relevant to a specific RFP clause requirement.
5. Be brief, but complete, in your responses.
6. Provide succinct, clear, and unambiguous responses; do not obfuscate your responses with unnecessary wordage.
7. Make sure to review and edit your proposal before submission.
8. All proposals are due December 1, 2006. Late submissions and/or revisions to submitted proposals may not be accepted. Deadline extensions may be granted under acceptable circumstances, only.

Proposal Evaluation

The proposals to the RFP will be judged on the following factors:

1. Satisfaction of system performance requirements
2. Price of the proposed solution
3. Adherence to each of the above general proposal guidelines
Important submission requirements:

- Submit Part 1 System Performance Requirements responses in MS Office WORD file format, excluding responses to RFP Clauses specifying PowerPoint format, e.g., Clause 1.0.1. When PowerPoint format is requested do not copy/paste PDF format graphics or images.
- Submit Part 2 System Pricing responses in MS EXCEL file format
PART 1: System Performance Requirements

Submit Part 1 responses in MS Office WORD file format except when otherwise noted.

1.0.0 System Overview

The VoiceCon Company plans to install a new IP Telephony System (IPTS) network to support its newly constructed Headquarters (HQ) facility, a Regional Office (RO), and three Satellite Branches (SBs) with Survivable Remote Gateway (SRG) capabilities.

Dedicated local IPTS call telephony servers must be installed at the HQ and RO facilities. All proposed call telephony servers must independently support all generic software features for the proposed IPTS model(s) as required in Section 5 of this RFP. The three SBs will be configured as survivable remotes behind the HQ IPTS call server with local trunk circuit services (Note: Survivability requirements for the SB facilities are identified later in this section). The proposed IPTS network solution may include a single fully distributed IPTS or no more than two IPTSs (each housed at HQ and RO facilities). If a single IPTS is proposed the distributed call servers must function and operate independently of each other, and support all generic software features as required in Section 5 of this RFP.

The HQ IPTS call server will initially support 1,360 station users at the HQ and three SB facilities. The RO IPTS call server will initially support 250 station users. See Figure 1 for an overview of the VoiceCon IPTS network. See Figures 2 – 6 for port capacity requirements at each of the five VoiceCon facilities.

VoiceCon anticipates 20% station user growth at the HQ and RO facilities, only, and the proposed IPTS network solution must accommodate this growth without replacement of any installed hardware/software. There is no anticipated growth at the SB facilities. A centralized messaging system will be housed at the HQ facility and must be capable of supporting station users located at all VoiceCon facilities (HQ, RO, and SBs).
Figure 1
Voicecon IPTS Network

HQ IPTS
1200 stations

SB1 SRG
100 Stations

SB2 SRG
50 Stations

SB3 SRG
10 stations

RO IPTS
250 Stations

WAN

HQ: Headquarters
RO: Regional Office
SB: Satellite Branch
IPTS: IP Telephony System
SRG: Survivable Remote Gateway
Figure 2
HQ Port Requirements

HQ IPTS
1200 stations
6 Local T1 circuits
7 Long Distance T1 circuits
5 PFTS circuits
25 Emergency Analog GS/LS Circuits

Figure 3
RO Port Requirements

RO IPTS
250 stations
2 Local T1 circuits
2 Long Distance T1 circuits
2 PFTS circuits
10 Emergency Analog GS/LS Circuits
Figure 4
SB1 Port Requirements

SB1 SRG
100 stations
1 Local T1 circuit
2 PFTS circuits
5 Emergency Analog Circuits

Figure 5
SB2 Port Requirements

SB2 SRG
50 stations
10 Analog LS/GS circuits
2 PFTS circuits
VoiceCon has plans to install at all of facilities LAN/WAN cabling and a transport infrastructure that will fully satisfy the stringent requirements of IP Telephony communications for all intra-premises and inter-premises call control and voice communications transmissions. Each location will be equipped, at minimum, with a 1-Gbps Ethernet backbone. The local wiring closets will house 10/100/1000 Mbps Ethernet switches equipped with Power over Ethernet (PoE). Multi-service routers will be installed at all locations to support a MPLS WAN installation. All Ethernet switches and IP WAN routers will be equipped and programmed to satisfy QoS and security standards necessary to support voice communications acceptable to VoiceCon. Pertinent bandwidth, latency, packet loss, and echo issues will be addressed in the design and implementation.

Each station user’s work area will be supported by four (4) four-pair, Category 5E cable wiring with one (1) RJ-11 wall connector and three (3) RJ-45 wall connectors to the local wiring closet. The RJ-11 and RJ-45 connectors will be either wall mounted or mounted in the modular furniture throughout the office environment. VoiceCon plans to run its IP Telephony system over this cable infrastructure. NOTE: The proposed IP Telephony system must be able to support a limited number of non-IP stations, e.g., analog telephones, requiring a RJ-11 connector. The proposed system can use either circuit switched port carriers or media gateways to support analog communications terminal equipment.
Vendor Response Requirement

Based on the RFP requirements in this document prepare a simple network diagram that illustrates the proposed IPTS network design. Include in the diagram the brand name/model of the IPTSs, all circuit switched port carrier/media gateway equipment, the brand/name of the HQ-located systems management and messaging system. The diagram must be prepared and submitted in MS PowerPoint format (identify the file as part of your electronic proposal submission), and also copy/paste here the diagram in the submitted MS WORD file proposal.

Proposed IPTS Network Diagram Here

1.0.1 LAN/WAN Requirements

VoiceCon has not yet decided on the make/manufacturer of its new LAN/WAN communications equipment.

Vendor Response Requirement

Indicate if the proposed IPTS solution for the HQ and the remote facilities requires manufacturer-specific LAN/WAN communications equipment to support any or all of the following voice communications operations or functions: call processing, switching, routing, PoE, media gateway, QoS and security. If responding in the affirmative, only, identify the make and model of the necessary switch/router equipment and the reason for its requirement.
SIEMENS RESPONSE:

The HiPath 8000 solution provides an open standards solution and does not require any manufacturer-specific LAN/WAN communications equipment.

1.1.0 Basic IPTS Requirements

The proposed IPTS equipment should be in current production and operating as part of a commercial system for at least five (5) customers in the USA.

Vendor Response Requirement
State if the proposed IPTS equipment satisfies this commercial availability requirement. If the IPTS model has not yet been shipped and installed in a commercial installation, state expected availability date. Also provide an estimate of the number of IPTSs (same model as proposed) currently installed and operating in the USA.

SIEMENS RESPONSE:

The HiPath 8000 version 2.2 is commercially available in the US and globally. Siemens has installed 4 systems in the US and over 20 systems in Europe. The installed version of software ranges from version 2.0 to version 2.2.

NOTE: All proposed system hardware and software must be formally announced as of VoiceCon Spring 2006 to be accepted by VoiceCon in response to this RFP. This is a mandatory requirement to submit a RFP response.
1.1.1 Single System Image

The proposed IPTS network should provide a Single System Image across VoiceCon HQ, RO and SB facilities. The Single System Image should include, but not be limited to, the following:

1) 5-digit dialing between all station users;
2) High degree of transparent operation across all VoiceCon facilities for station, attendant, and system features (see RFP Section 5: Call Processing Features);
3) HQ-located centralized systems management solution using a single unified database for all station user profiles, equipped system design, and system-level operations;
4) Network-wide attendant operator services across all VoiceCon facilities, including the ability to support a centrally located attendant pool;
5) Shared messaging system resources;
6) Automatic alternative routing across the network for all voice calls (station-to-station and PSTN trunk connections).

Vendor Response Requirement:
Provide specific answers to each of the following questions:
1. Is the proposed IPTS network solution a single system solution or multiple systems intelligently networked?
2. Does the proposed IPTS network solution fully satisfy all six (6) of the stated Single System Image requirements? If not, explain which of the requirements are not satisfied?

SIEMENS RESPONSE:

The solution proposed is a single solution using the HiPath 8000 and survivable gateways at the remote locations. The HiPath 8000 meets all of the requirements above with the exception of number 3. Currently the HiPath 8000 Assistant provides for management of the HiPath 8000 and the RG 8700 series gateways. The Voice Mail system and the other remote gateways are not managed by the HiPath 8000 Assistant. The Xpressions system has its own centralize management system with would be co-located with the HiPath 8000 assistant. The Mediatrix and Comdasys gateways are managed at the site.

1.1.2 Enhanced 911 (E911) Services Support

It is mandatory that the proposed and installed communications system support E911 services provided by a public safety answering point (PSAP) as defined by FCC regulations. All station user E911 calls must be directed to their local PSAP for call handling and response regardless of location, i.e., facilities remote from the primary call telephony server. If more than one E911 solution is available for the proposed IPTS network configuration clearly specify the solution that is included in the price proposal.
**Vendor Response Requirement:**
Confirm that the proposed communications system solution supports E911 service for all user stations (IP and analog) at each of the VoiceCon facilities. In the response briefly explain how E911 service requirements are supported, specifically addressing each of the following questions:

1) A description of any optional hardware/software equipment included in the pricing proposal, and if a peripheral server is required who is responsible for its purchase?

2) How are station user moves/adds/changes reported to the E911 provider?

3) What degree of specificity station user location is identified to the E911 PSAP? Desktop work area, local switch room, work floor, other?

**SIEMENS RESPONSE:**

The HiPath 8000 associates to a LIN based on calling party SIP Invite IP address. The HiPath 8000 maintains an E911 table with records containing IP address ranges, an associated LIN, and an associated route ID(gateway route to place the call to the appropriate PSAP). To control the IP address DHCP distribution at a more granular level than a subnet\ VLAN, Siemens recommends the customer deploy DHCP Relay Agent, DHCP option 82. Today, Siemens uses a channel bank and analog CAMA trunks to send the E911 information to the PSAP. Siemens is currently working on a gateway solution to send the LIN information in the CPN field.

Station Moves do not need to be reported to the service provider, since our E911 strategy is built upon the IP infrastructure. LIN information needs to be managed with the service provider and correlated with the customer's IP addressing scheme.

**1.1.2.1 E911 and Station Moves**

It is desirable, but not mandatory, that station user moves behind the proposed IPTS solution be tracked dynamically in real time for E911 services support.

**Vendor Response Requirement:**

Indicate if the proposed E911 solution satisfies this desirable capability and indicate how often the database updated. If an alternative E911 solution is available that satisfies this capability, but is not included as part of the overall IPTS solution and pricing proposal, briefly describe this option and the incremental costs to purchase and install beyond the proposed solution.

**SIEMENS RESPONSE:** The HiPath 8000 Assistant updates the E-911 information whenever a database change is made. There is no special equipment needed to provide this function.
1.2.0 Proposed Communications System Design

The proposed communications system may only be based on either of the two following architecture technology designs:

- **Single system design** based on true peer-to-peer distributed call processing topology, i.e., identical or similar call telephony servers located at all VoiceCon facilities (HQ, RO, SBs)
- **Intelligently networked multiple system design** based on identical or similar call telephony servers located at VoiceCon HQ and RO facilities, and survivable remote gateways at VoiceCon SB facilities configurable behind the HQ call telephony server.

Only a supplier’s most current generation hardware/software solution will be acceptable. No refurbished equipment is acceptable.

**NOTE**: There is no preference for either the single or multiple system design if all **1.1.1 Single System Image** requirements are satisfied.

**Vendor Response Requirement**:
Briefly describe your proposed solution, referring to the diagram from RFP Clause 1.0.0 when applicable.

**Limit** your response in this section to the following high level information as details are requested in following sections:

1. Product and model name(s) for the IPTS(s) and messaging system.
2. Identify proposed solution as a single system or multiple system design.
3. For each network location specify the product/model used to support station/trunk call processing and switching operations under normal operating conditions.
4. Identity the software release for each product/model proposed
5. Provide the product/model introduction dates.

**SIEMENS RESPONSE**:

Siemens is proposing a dual node HiPath 8000 for VoiceCon. The HiPath 8000 as proposed would have one node located at the Voice Con Headquarters and the second node located at the regional office. This configuration allows for fully survivability in case the WAN link to the headquarters is lost. The configuration also includes RG8700 series gateways at the headquarters and at the regional office. These gateways are also survivable in case of loss of the HiPath 8000 in a catastrophic event. The other remote locations are equipped with Mediatrix trunking gateways and Comdasys survivable media proxies. Each site is capable of operation if
connection to the headquarters site and the regional office site is lost.

Proposed Equipment
HiPath 8000 dual node system, version 2.2 – introduction date September 2006
This includes:
Dual IBM 3650 servers – introduction date December 2006. Can use IBM X346 servers which are currently in production.
HiPath 8000 version 2.2 software

Gateways proposed
A single RG8716 at the headquarters site
A single RG8708 at the regional office site
A single Mediatrix T’1 trunking gateway and a Comdasys 1600 at the SB! SRG site
Three Mediatrix Analog Trunking gateways and a single Comdasys 150 at the SB2 SRG site.

1.3.0 System Design Platform

The proposed system solution may be based on either of the following two architecture system design:

- Converged TDM/IP: call telephony server supporting LAN/WAN distributed circuit switched port interface cabinets with equipped media gateway interfaces for IP port connectivity
- Client/server: call telephony server supporting media gateway equipment (server-embedded, standalone, switch/router-equipped or desktop) for non-IP port connectivity

Vendor Response Requirement:
Briefly and clearly describe the architecture and design elements of the proposed IPTS solution. Include in your basic system description information about the following common equipment hardware elements:

1. Type of architecture design (converged or client/server)
2. Call telephony server and associated common control equipment
3. If applicable, circuit switched port interface equipment housing TDM port interface circuit cards and media gateway boards.
4. If applicable, LAN-connected media gateways (server-embedded, standalone, switch/router-equipped, desktop

SIEMENS RESPONSE:

The HiPath 8000 provides a client/server architecture. The HiPath 8000 provides common control via a pair of IBM X3650 e-Servers. Additionally the
HiPath 8000 uses an IP Unity Media server to provide tones and announcements and for integration of other applications. The HiPath 8000 does not use circuit switched port interface equipment.

1.3.1 Common Control

The primary common control complex of the proposed IPTS should be based on a standalone call telephony server or a call processor blade that is embedded in common equipment that functions as a call telephony server. The physical equipment may either be a fully bundled proprietary hardware/software offering that is factory configured or third party equipment provided by VoiceCon that is capable of running proposed proprietary call processing software without any service degradation.

Any and all of the proposed primary common control call processor elements used to provide call processing functions must be proposed in a redundant duplicated design with seamless switchover operation between active and standby control elements, i.e., all active call connections must remain up during switchover in case of failure or major alarm states and new calls set-up without delay. The common control design may be based on a load sharing design in which any call telephony server/processor blade may be programmed to function in primary and secondary backup modes. All common control elements must be capable of supporting required equipped and wired capacities at time of installation.

The call processing rating for all proposed IPTS call servers must minimally support 50,000 Busy Hour Call Completions (BHCCs).

Vendor Response Requirement: Provide a brief description of the common control design you are proposing in terms of design platform: call telephony server, call processor blade (including necessary housing), third party server. Confirm that the duplicated common control requirement is fully satisfied by the proposed solution, and identify any feature/function that is not available if a standby (back-up) call processing element must be activated in case of a primary element failure.

1.3.2 CPU Make/Model

Vendor Response Requirement: Identify the make/model of all proposed common control CPU(s) and associated BHCC rating for the configured system.

SIEMENS RESPONSE:

The HiPath 8000 uses dual Intel Xenon processors in each server. The HiPath 8000 supports 70,000 BHCCs.

1.3.3 Call Processing O/S
**Vendor Response Requirement:**
Identify the primary operating system of the common control call processor. A version of Linux is preferred, but not mandatory.

**SIEMENS RESPONSE:**

The primary operating system for the HiPath 8000 is SuSE Linux version 9.0.

**1.3.4 Memory**

**Vendor Response Requirement:**
Briefly describe the main memory design and storage elements and capacity for both the generic software and customer database as proposed.

**SIEMENS RESPONSE:** The HiPath 8000 uses dual 135 gigabit drives in each server. The generic software and the customer database are both stored in the drives.

**1.3.4.1 Database Integrity**

**Vendor Response Requirement:**
How does the proposed IPTS solution maintain the integrity of the customer database between back-ups?

**Siemens Response:** The HiPath 8000 Resilient Telco Platform (RTP) maintains the integrity of the database during normal operation. The database is stored in both IBM e-Servers and during normal operation the RTP continually monitors the database for errors. Any errors in a database are corrected by the RTP using the database in the partner server.

**1.3.4.2 Database Information Loss**

**Vendor Response Requirement:**
Identify under what circumstances can customer database information (configuration, messages, logs, etc.) be lost during back-ups

**Siemens Response:** Due to the dual node nature of the HiPath 8000 and the fact that both nodes house the entire database this has never occurred on a HiPath 8000.

**1.3.4.3 Database Backup Scheduling**

**Vendor Response Requirement:**
How often should the customer database be backed up? Specify if it is a full or incremental backup and the time the process takes.

Siemens Response: Siemens recommends that the HiPath 8000 be fully backed up on a monthly basis. The average time is 1.75 hours per node, on and idle system.
1.3.4.4 Data Purging/Archiving

Vendor Response Requirement:
Describe the mechanism for data purging and archival, including storage and retrieval of archived data.

Siemens Response:

The Linux file system back-ups can be kept on an external universal serial bus hard disk drive (USB HDD) or a back-up server. If a back-up server is used it must meet the following requirements:
- Supports FTP or sFTP for transfer of back-ups.
- Has adequate disk space for the back-ups.
- Stores at least two copies of each back-up on separate physical devices.
- Has a connection to the HiPath 8000 of adequate bandwidth to allow a fast recovery.

The back-up server may be dedicated to the HiPath 8000 or used for multiple devices depending on the back-up strategy. The back-up server must meet the following requirements:
- The server shall be capable of receiving and sending HiPath 8000 back-ups generated using the procedures described in Section 3.2, “Backup File System for a Simplex or Duplex System”, on page 3-6. Transfers will be via FTP or Secure FTP (OpenSS).
- The server will not store the HiPath 8000 database back-ups as these are kept on the iNMC.
- Redundancy is required, for example by using duplicated disk drives.
  – The server should have two disk partitions on different disk drives—primary back-up and secondary back-up.
  – All transfers to and from the HiPath 8000 will normally be to the primary back-up partition.
  – Manual operation or scripts can copy the back-ups from primary to secondary partitions.
- File management, manual or scripted, is required to:
  – Keep secondary back-up synchronized with primary back-up partition.
  – Maintain two back-ups per HiPath 8000. When a new back-up is received the oldest will be deleted.
  – Mark a back-up as persistent such that it will not be automatically deleted.
  – Delete old back-ups if the disk becomes full and transfer of back-ups from the HiPath 8000 to server is not possible.
- In place of duplicated disks the server may use:
  – An optional optical or tape back-up system.
  – Commercial disk shadowing mechanism.

Backup and Restore Guidelines
Back-up Procedures
• Use an existing back-up server, used for other equipment, to store the HiPath 8000 backups.
• Bandwidth of the connection between the HiPath 8000 and the server. The back-ups are large, hence low bandwidth will impact recovery time and prolong the outage.
• Should the server be local to the HiPath 8000. If local, off-site copies of the back-ups must be maintained:
  – By transfer (FTP) to a remote server.
  – Using back-up tape system. Tapes can then be held on and off-site.

1.3.5 Power Supply

**Vendor Response Requirement:**
Briefly describe common control power requirements and the integrated power distribution design. Indicate if the power supply is dependent on either an AC or DC current source.

Siemens Response: The HiPath 8000 uses industry standard IBM X346 or X3650 e-Servers. The power requirements at 110 or 220 V AC.

1.3.5.1 Power Safeguards

**Vendor Response Requirement:**
Describe any power failure safeguards that are included in the IPTS design. Briefly describe what happens to system operation during a power failure.

Siemens Response: Each IBM e-Server is equipped with dual power supplies. In case of a failure of one of the power supplies the secondary power supply will back up the first. In addition the design of the HiPath 8000 provides for full back up from the partner server in case of power failure in it’s partner server.

1.3.5.2 Power Backup

**Vendor Response Requirement:**
Is the proposed IPTS solution equipped with standard UPS hardware, and if so how long can the system run on it? If not, what UPS requirements are recommended?

Siemens Response: The HiPath 8000 as designed does not have UPS hardware. The type of UPS will depend on the requirements of VoiceCon and the length of time needed for back up in case of power failure. Note that all of the other locations continue to operate in case of power supply failure due to the survivable nature of the HiPath 8000 architecture.
1.3.6 Ethernet Call Control Signaling Links

Vendor Response Requirement:
Identify for each active and standby call telephony server the number of available and configured RJ-45 Ethernet LAN uplink interfaces for call control signaling to LAN-connected cabinets/carriers and/or standalone ports. Include a brief description of how the physical Ethernet connection is provided: dedicated circuit board; daughterboard; fully integrated RJ-45 connector, et al.

Siemens Response: The attached diagram should all of the connections with the HiPath 8000 dual servers. The HiPath 8000 is connected to the IP devices via dual Gigabit switches. The switches are usually customer provided and are connected via RJ45. Each server in the HiPath 8000 is Active/Active with its partner server. All uplink interfaces are located in the servers on PCI boards.

1.3.7 System Clocks

Vendor Response Requirement:
Identify the number and type of internal system clocks that are available and configured.

Siemens Response: Each processor in the IBM e-Series server has a clock that provides time to the system. Each IP device also has a clock to provide time of day, day of week, etc. Note that the HiPath 8000 gets this
information from the IP network.
1.3.8 Redundant system design elements

It is desirable to have a highly redundant system design, especially as it relates to common control elements necessary for call processing, maintenance, and administration operations.

Vendor Response Requirement:

Specify the level or degree of redundancy included in your proposal for each of the following listed common control elements. For example, full duplicated back-up, standby load sharing, seamless switchover, cold standby, et al.

- Primary call processor
- Main system memory
- Customer database memory
- RJ-45 Ethernet uplinks to network
- Power supply
- Tone generators
- Call classifiers
- Registers
- DTMF receivers
- I/O interfaces

Siemens Response: The HiPath 8000 provides full redundancy for all items listed above. Based on the dual server active/active design of the HiPath 8000 all items are fully redundant. During the failure of any of the components above the partner server takes over the processing of the calls. Both servers contain main memory, customer database and RJ-45 uplinks. Both servers are designed with dual power supplies and all other interfaces are duplicated as well.

1.4 Local Survivability

It is important to VoiceCon that station users at all network locations have access to telephony services at all times. This includes 100% of generic software features and trunk circuit access to a local exchange carrier. For this reason it is highly desirable that station users at VoiceCon’s SB facilities have access to telephony services in case of HQ-SB WAN link failure due to switch, router, or private network transmission service issues, or HQ common control failure for any reason.

It is preferable that standby telephony services be provided by an on-site call processing option. A less desirable, but acceptable, emergency option is an alternative PSTN-based call control signaling link, but only if an on-site call processing option is not available as part of the system solution. It is also highly desirable that the standby call processing option provide stations users
with the **same level of telephony services, e.g., station, attendant, and system features**, supported by the HQ IPTS at the medium (50 stations) and large (100 stations) SB facilities. For the small (10 stations) SB facility it is acceptable that POTS-like survivability (dial tone, PSTN trunk access, intercom calls) is supported. **SB facility local survivability for any disruption of HQ-based IPTS call control signaling is a mandatory requirement for proposal submission.**
Vendor Response Requirement:
Describe the proposed local survivability solution that satisfies the stated requirements. Include a description of all proposed local survivability options (including any and all required hardware, software, and PSTN transmission services to implement the option) for each of the three SB facilities: 10 stations, 50 stations, 100 stations.

Siemens Response: The Siemens solution provides for a dual node HiPath 8000 with the nodes located at both the HQ and RO location. This allows for the WAN link at the HQ location to be lost and for all other users to continue to have fully HiPath 8000 services. If both the WAN link to the HQ and RO location are lost then all other locations are survivable via the included survivable media gateways. The HiPath 8000 will switch to the remaining node automatically if the WAN link to the HQ is lost. If both the line to the HQ and the RO are lost then the local survivable media gateways will switch in automatically. All locations will have independent trunking and therefore will be able to make and receive calls.

1.4.1 Survivable IPTS Features/Services

Vendor Response Requirements:
Identify any required generic software feature (See Section 5.0 Call Processing Features) not available or operational when the local survivability solution is activated at VoiceCon’s 50 station and 100 station SB facilities. Also identify any station user equipment not supported in standby survivability mode at these two facilities.

Siemens Response: The features listed below are available in the survivable mode to the SB facilities. These are the only features that are available in the survivable mode.

SIP Features available: (Generally, local phone/client features will still function)
- SIP to SIP calls. SIP to GW calls. GW to SIP calls.
- Only Prime Line can be used.
- Digest Authentication of Client.
- Local 3-party conference calls.
- In-Band DTMF generation.
- Local Forwarding
- Do Not Disturb
- Headset operation
- open-listening
- speakerphone
- mute
- Missed Call List
- Repeat Dial List
- Abbreviated Dialing List
- Ringer on/off function
- Key Click on/off
- Tone Generation
- Local Display of Time of Day Clock (unsynchronized)
- Call Duration time Display
- Repertory Dialing (assuming number is reachable)
- Drop Call Key
- Message Waiting displayed in idle menu
- Local ID shown in idle menu
- Display of Customer logo/name.
- Hold
- Consultation/Alternate
- Attended & Unattended transfer
- Music On Hold & Call Park (only if local functions)
- Directed Call Pickup.

Table 9 Station User Features - The following station features are not available in Survivable Mode

STATION USER FEATURES

ADD-ON CONFERENCE (6 party or more)
AUTOMATIC CALLBACK
AUTOMATIC INTERCOM
BRIDGED CALL APPEARANCE
CALLBACK LAST INTERNAL CALLER
CALL COVERAGE (PROGRAMMED)
CALL FORWARDING - ALL CALLS
CALL FORWARDING - BUSY/DON'T ANSWER
CALL FORWARDING - FOLLOW-ME
CALL FORWARDING - OFF-PREMISES
CALL FORWARDING: RINGING
CALL PARK
CALL PICKUP – GROUP
CONSECUTIVE SPEED DIALING
CUSTOMER STATION REARRANGEMENT
DIAL BY NAME
DISCRETE CALL OBSERVING
ELAPSED CALL TIMER
EMERGENCY ACCESS TO ATTENDANT
EXECUTIVE ACCESS OVERRIDE
EXECUTIVE BUSY OVERRIDE
FACILITY BUSY INDICATION
GROUP LISTENING
HELP INFORMATION ACCESS
HOT LINE
INDIVIDUAL ATTENDANT ACCESS
LINE LOCKOUT
LOUDSPEAKER PAGING ACCESS
MALICIOUS CALL TRACE
MANUAL ORIGINATING LINE SERVICE
MEET ME CONFERENCING (6-Party or more)
MULTI-PARTY ASSISTED CONFERENCE w/SELECTIVE CALL DROP
OFF-HOOK ALARM
PADLOCK
PAGING/CODE CALL ACCESS
PERSONAL CO LINE (PRIVATE LINE)
PRIORITY CALLING
PRIVACY - ATTENDANT LOCKOUT
PRIVACY - MANUAL EXCLUSION
SECONDARY EXTENSION FEATURE ACTIVATION
SEND ALL CALLS
SILENT MONITORING
STEP CALL
SUPERVISOR/ASSISTANT CALLING
SUPERVISOR/ASSISTANT SPEED DIAL
TEXT MESSAGES
TIMED QUEUE
WHISPER PAGE

Table 11 System Features – The following System Features are not available in Survivability Mode.

SYSTEM FEATURES

ACCOUNT CODES
ADMINISTERED CONNECTIONS
AUTHORIZATION CODES
AUTOMATED ATTENDANT
AUTOMATIC CALL DISTRIBUTION
AUTOMATIC ALTERNATE ROUTING
AUTOMATIC CAMP-ON
AUTOMATIC CIRCUIT ASSURANCE
AUTOMATIC NUMBER ID
AUTOMATIC RECALL
AUTOMATIC ROUTE SELECTION – BASIC
AUTOMATIC TRANSMISSION MEASUREMENT SYSTEM
CALL-BY-CALL SERVICE SELECTION
CALL LOG
CENTRALIZED ATTENDANT SERVICE
CLASSES OF RESTRICTION (SPECIFY #)
CLASSES OF SERVICE (SPECIFY #)
CODE CALLING ACCESS
CONTROLLED PRIVATE CALLS
DIALED NUMBER ID SERVICE
DIRECT DEPARTMENT CALLING
DIRECT INWARD SYSTEM ACCESS
DIRECT INWARD TERMINATION
EXTENDED TRUNK ACCESS
FACILITY RESTRICTION LEVELS
FACILITY TEST CALLS
FIND ME- FOLLOW ME
FORCED ENTRY ACCOUNT CODES
HOTELING (PERSONAL ROAMING)
HOUSE PHONE
HUNTING
INTEGRATED SYSTEM DIRECTORY
LEAST COST ROUTING (Tariff-based, TOD/DOW)
  MULTIPLE LISTED DIRECTORY NUMBERS
  MUSIC ON HOLD
  NIGHT SERVICE – FIXED
  NIGHT SERVICE – PROGRAMMABLE
  OFF-HOOK ALARM
  OFF-PREMISES STATION (OPX)
  OPEN SYSTEM SPEED DIAL
  PASSWORD AGING
  POWER FAILURE TRANSFER STATION
  RECENT CHANGE HISTORY
  RESTRICTION FEATURES
    CONTROLLED
    FULLY RESTRICTED
    INWARD/OUTWARD
    MISCELLANEOUS TERMINAL
    MISCELLANEOUS TRUNK
    TOLL/CODE
    TRUNK
    VOICE TERMINAL (IN/OUT)
ROUTE ADVANCE
SECURITY VIOLATION NOTIFICATION
SHARED TENANT SERVICE
SNMP SUPPORT
SYSTEM SPEED DIAL
SYSTEM STATUS REPORT
TIME OF DAY ROUTING
TIMED REMINDER
TRUNK ANSWER ANY STATION
TRUNK CALLBACK QUEUING
UNIFORM CALL DISTRIBUTION
UNIFORM DIAL PLAN
VIRTUAL EXTENSION
VOICE MESSAGE SYSTEM INTERFACE
1.4.2  Local Survivability Failover and Switchback

Vendor Response Requirements:
For each of the SB facilities is failover to the local survivable call processing option seamless, i.e. no interruption of in-process telephony services, for any or all stations users if WAN connectivity is disrupted to the HQ IPTS? Indicate in answer if there is delay for implementing new calls immediately after the WAN disruption. Also describe the switchback process when HQ facility IPTS call control is again available via the WAN, specifying if the process is automatic or manual and how long the process takes to implement. Are connected calls and voice operations at the remote facility affected in any way by the switchback process and how soon can new calls be implemented?

Siemens Response: The HiPath 8000 and the RG 8700 series survivable media gateway use dual registration. The RG8700 gateway sends out a SIP invite to the HiPath 8000 is the invite is not returned the RG8700 automatically switches to survivable mode. All users on the gateway are switched automatically and there is no interruption of either internal or external calls. For sites using the Mediatrix/Comdasys solution the users are registered to the Comdasys solution via a SIP Proxy. The Comdasys solution registers to the HiPath 8000. If connection is lost the users continue to have service provided.

1.4.3 Survivable Messaging Services

It is desirable that remote station users at the RO and SB facilities have access to messaging services if there is a WAN link disruption to the HQ messaging system.

Vendor Response Requirements
Does the proposed IPTS network and messaging solution satisfy this requirement if WAN connectivity between HQ and any of the other facilities (RO, SBs) is not available? Briefly describe how messaging services would be implemented and accessed by remote station users in emergency situations. The minimum messaging services function in survivable mode should include voice mailbox access by station users.

Siemens Response: Exception. The Siemens Xpressions system as designed does not meet this requirement, however, the Xpressions system can be designed to provide survivability with an optional secondary processor at the Regional Office.
1.4.4 Network Failover Resiliency

In the unlikely event the redundant common control complex (primary active and secondary backup) at either the HQ or RO facilities become nonfunctional due to extreme system failure or catastrophic circumstances, e.g., fire, VoiceCon requires implementation of a resilient network failover process. This process requires that all local station users and media gateway equipment configured behind the nonworking common control complex automatically re-register to a designated emergency call telephony server(s) at either the local or remote facility for continuity of telephony services. For this reason it is necessary that the designated emergency call telephony server(s) located at the HQ/RO facility be capable of supporting sufficient port capacity requirements in the event of a failover.

Vendor Response Requirements
Does the proposed IPTS solution support network failover resiliency in case of a catastrophic common control failure at either the HQ or RO facilities? If affirmative, describe the failover process, optional hardware/software and/or WAN transmission requirements to implement, and the time required for the network failover to be implemented before telephony services are available. Indicate if the proposed IPTS solution can support more than one network failover design.

Siemens Response: As proposed the HiPath 8000 system is using a split node architecture that allows for either the HQ or RO facilities to be in a failure situation. This design provides for a layer 2 connection between the nodes of the HiPath 8000. If the connection between the two nodes is severed each individual node can operate the entire system with no loss of calls. The Resilient Telco Platform middleware provides for an intelligent monitoring of this connection. Because all call handling and database information is stored in both nodes at all times all information is constantly available to both nodes of the system. There is no failover time when this feature is activated. In addition the proposed solution provides for the use of Survivable Media Gateways to ensure survivability of all remaining locations even if both nodes of the HiPath 8000 are lost or unable to be contacted.

1.5 Session Initiated Protocol (SIP)

VoiceCon requires that the proposed IPTS support SIP-compatible stations and trunk networking. It is also required that the IPTS solution be capable of supporting the IETF-sponsored signaling protocol used for Internet conferencing, telephony services and features, presence, events notification and instant messaging.
1.5.1 SIP Stations

Vendor Response Requirements
Indicate if the IPTS solution as proposed can currently support SIP-compatible desktop telephone instruments (self and/or third party) and PC client softphones. Specify if SIP call control is embedded in the IPTS common control design or requires optional hardware/software elements. Please identify up to three (3) third party SIP telephones you have tested to work behind your proposed IPTS solution.

SIEMENS RESPONSE:

The HiPath 8000 is a SIP overlay system that has been designed specifically to provide communications using the SIP protocol it is embedded in the IPTS common controls design and does not require options hardware or software. Siemens has used open standards and can use other RFC3261 compliant devices. Other manufacturers phones that can be used on the HiPath 8000 include the Cisco 7940 and 7960 phones, Grandstream phones and Polycom phones.
1.5.1.1 SIP Clients

Vendor Response Requirements
Do the IP desktop telephone instruments and PC client softphones included as part of this proposal in response to RFP Section 5: Voice Terminals currently conform to IETF SIP standards? If not, are they upgradeable via a firmware download if required in the future?

SIEMENS RESPONSE:

The phones proposed conform to the IETF SIP standards.

1.5.2 SIP Trunk Networking

Vendor Response Requirements
Indicate if the IPTS network solution as proposed can support SIP-based trunk networking. Specify if SIP media proxies are required to support this requirement. Identify up to three (3) major Service Providers (SPs) and three (3) other IPTS suppliers you have conducted IP trunk networking compatibility tests with for the proposed IPTS.

Siemens Response: The HiPath 8000 and the RG8700 are designed to support SIP trunking. At this time Siemens has not tested with any network provider or Service Provider. This is scheduled for early 2007.

1.5.2.1 SIP Applications

Vendor Response Requirements:
Indicate if the IPTS network solution as proposed can support SIP-enabled applications, such as Internet conferencing, telephony services and features, presence, events notification and instant messaging.

SIEMENS RESPONSE: The HiPath 8000 supports SIP-enabled applications. Currently the HiPath 8000 supports the following SIP-enabled applications

HiPath Xpressions – Unified Messaging
IP Unity UM
IP Unity Conferencing
Gensys Call Center
HiPath Pro-Center version 7.0.

1.6 Security
VoiceCon requires a secure IPTS network solution to optimize system performance and reduce the probability of toll fraud and illegal system access.

1.6.1 Authentication

Vendor Response Requirements
Briefly describe authentication processes embedded in the proposed IPTS solution to prevent: unauthorized access to common control elements, data resources; and abuse of telephony services, e.g., toll fraud.

Siemens Response

The diagram above indicates how the HiPath 8000 provides security for the common control elements. The HiPath 8000 uses TLS and X.509 certificates as well as digest authentication to ensure that the end users are properly registered. Specifically the HiPath 8000 is fully RFC 2246 and 3261 compliant.

1.6.2 Disruption of Services

Vendor Response Requirements
Briefly describe any embedded features/functions in the proposed IPTS that will reduce probability of telephony services disruption due to Denial-of-
Service (DoS) attacks.

Siemens Response: The HiPath 8000 protects against DoS attacks in the following manner.

- **Message Filtering:**
  - H8k accepts packets only from relevant IP addresses and ports. All other traffic is denied.

- **Message Throttling (using built-in Snort):**
  - Rate checking applied against each packet source (IP address)
  - IP addresses sending excessive packets are blacklisted (Restored to white list after timed period)
  - Separate thresholds for servers and endpoints
  - DoS Alarming

- **Codenomican Test Suite run on every release:**
  - PROTOS Test-Suite:c07-sip
  - IETF SIP Torture Test
1.6.3 Confidentiality and Privacy (Packet Sniffing)

Vendor Response Requirements
Briefly described any embedded features/functions in the proposed IPTS that will preserve communications confidentiality and privacy. Indicate if control signaling and/or bearer communications signaling is encrypted at the call control, voice client, and media gateway elements to counter packet sniffing attempts.

Siemens Response: The HiPath 8000 and Siemens end points support RFC 3711 and 3830. The information below describes the current and future embedded functions to provide this protection.

HiPath 8000 supports:
- Step 1
  - SRTP support for SIP station-to-station connections
  - OptiPoint 410/420 V7 support of SRTP, OpenStage
  - Key management support for MIKEY 0 or SDesc
    - TLS must be used to secure SIP signaling connection
- Step 2 (planned V3.0/V3.1)
  - SRTP sync point for all H8k components (MS, GWs, etc.)
  - Interworking support of CALEA/LI

1.6.4 Physical Interfaces

Vendor Response Requirements
Are there separate physical network interfaces to IPTS administration, control, and voice transmission signaling functions?

Siemens Response: The HiPath 8000 uses three separate sub-nets for the administration, call control and voice transmission.

1.6.5 Root Access

Vendor Response Requirements
Is there direct Root access to the IPTS common control?
Siemens Response: No. Access to the IPTS common control is provided by the HiPath 8000 Assistant.

2.0 IPTS Network Port Capacity Requirements

The proposed IPTS must be capable of supporting port capacity requirements for the HQ facility and remote branches. It must also be capable of supporting future VoiceCon growth requirements at HQ and RO facilities.

2.1.0 Port Capacity Requirements

The equipped port capacity of the proposed VoiceCon HQ IP Telephony System at time of installation and cutover must support of a mix of IP telephones, analog telephones, facsimile terminals, modems, local central office trunk circuits (analog and digital, long distance trunk circuits [digital, only], and private network trunk circuits [IP]).

In support of general communications requirements, VoiceCon facilities will have a sufficient number of wiring closets distributed throughout each facility to satisfy ANSI/EAI/TIA 569 structured cabling specifications for voice and data communications. Wiring closets will be interconnected based on requirements of the selected system. The entrance facility (trunk connect panel), main telecom equipment room, and Main Distribution Frame (MDF) for each facility are located off the entrance lobby. It will be the responsibility of the contractor to provide all cross connects between labeled 110 terminal blocks in each wiring closet and the demarc or "smart jack" and their equipment.
The following describes the port capacity requirements for each of the VoiceCon network locations. Satisfying these stated port capacity requirements is a **MANDATORY** requirement.

### 2.1.1 HQ Facility

The HQ location is a four-floor facility that will support at time of system installation and cutover the following station equipment:
- **1110** desktop IP stations: **1000** desktop instruments, **100** PC client softphones (including three attendant operator positions), and **10** audio conferencing units;
- **26** analog telephones including **5** used for Power Failure Transfer Station operation;
- **12** facsimile terminals;
- **12** data modems.

Station equipment is uniformly distributed within and across the four floors of the building. There are ten (10) wiring closets per floor, and one (1) main equipment room on the first floor. See Table 1 for station summary.

**Siemens Response: Siemens has read and will comply.**

### 2.1.2 RO Facility

The RO facility will be a two floor facility that will support:
- **230** desktop IP stations: **205** desktop instruments, **21** PC client softphones (including two (2) attendant operator positions), and **4** audio conferencing units;
- **10** analog telephones including **2** used for Power Failure Transfer Station operation;
- **5** facsimile terminals;
- **5** data modems.

Station equipment is uniformly distributed within and across the two floors of the building. There are ten (10) wiring closets per floor, and one (1) main equipment room on the first floor. See Table 1 for station summary.

**Siemens Response: Siemens has read and will comply.**

### 2.1.3 SB1 (Large)

The SB1 facility will be a single floor facility that will support:
- **89** IP stations: 75 desktop instruments, 10 PC client softphones, and 4 audioconferencing units;
- **5** analog telephones including 2 Power Failure Transfer Stations;
- **2** facsimile terminals;
2.1.4 SB2 (Medium)

The SB1 facility will be a single floor facility that will support:
* 46 IP stations: 37 desktop instruments, 5 PC client softphones, and 4 audioconferencing units;
* 2 Analog station used as Power Failure Transfer Stations;
* 2 facsimile terminals;
* 2 modems.

All line station equipment will be equally distributed across the single floor of the building. There will be two (2) wiring closets, and one (1) main equipment room. See Table 1 for station summary.

Siemens Response: Siemens has read and will comply.

2.1.5 SB3 (Small)

The SB3 facility will be a single floor facility that will support:
* 8 Desktop IP station instruments;
* 1 Analog station used as a Power Failure Transfer Station;
* 1 facsimile terminal.

All station equipment will be equally distributed across a single room on the main floor of the building. There will be one (1) wiring closet/equipment room. See Table 1 for station summary.

Siemens Response: Siemens has read and will comply.

<table>
<thead>
<tr>
<th>Table 1: VoiceCon Equipped Station Requirements</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP Station</td>
</tr>
<tr>
<td>------------</td>
</tr>
<tr>
<td>HQ</td>
</tr>
<tr>
<td>RO</td>
</tr>
<tr>
<td>SB1</td>
</tr>
<tr>
<td>SB2</td>
</tr>
<tr>
<td>SB3</td>
</tr>
</tbody>
</table>
2.2 Equipped Voice Terminal Requirements

VoiceCon requires the following mix of wired and installed desktop IP telephone instruments (Table 4). Note: Descriptions of Desktop IP individual voice terminal types can be found in RFP Section 4

Table 4: VoiceCon Desktop IP Telephone Instruments Requirements

<table>
<thead>
<tr>
<th>Facility</th>
<th>Economy</th>
<th>Administrative</th>
<th>Professional</th>
<th>Executive</th>
</tr>
</thead>
<tbody>
<tr>
<td>HQ</td>
<td>50</td>
<td>140</td>
<td>800</td>
<td>50</td>
</tr>
<tr>
<td>RO</td>
<td>15</td>
<td>30</td>
<td>150</td>
<td>10</td>
</tr>
<tr>
<td>SB1</td>
<td>8</td>
<td>10</td>
<td>55</td>
<td>2</td>
</tr>
<tr>
<td>SB1</td>
<td>3</td>
<td>8</td>
<td>25</td>
<td>1</td>
</tr>
<tr>
<td>SB3</td>
<td>0</td>
<td>1</td>
<td>7</td>
<td>0</td>
</tr>
</tbody>
</table>

2.3 Trunk Circuit Requirements

The VoiceCon HQ and RO facilities will each have a combination of local, long distance, and private network trunk circuits. The SB facilities will each have a limited number of analog trunk circuits, but all long distance calls will be routed through the HQ facility. All facilities will also have PFTS circuits. All local digital trunks must be able to support a combination of inbound DID service and two-way CO trunk services. All long distance calls placed from a SB facility will be routed via the LAN/WAN through the HQ facility for PSTN trunk access.

The following table summarizes HQ facility trunk circuit requirements for each of the four VoiceCon design configurations.
Table 5: VoiceCon IPTS Network Equipped Trunk Port Requirements

<table>
<thead>
<tr>
<th>Per Incremental Location</th>
<th>T-1 Digital Local Inbound/Outbound</th>
<th>T-1 Digital Long Distance</th>
<th>Analog (PFTS)</th>
<th>2-way GS/LS</th>
</tr>
</thead>
<tbody>
<tr>
<td>HQ</td>
<td>6</td>
<td>7</td>
<td>5</td>
<td>25</td>
</tr>
<tr>
<td>RO</td>
<td>2</td>
<td>2</td>
<td>2</td>
<td>10</td>
</tr>
<tr>
<td>SB1</td>
<td>1</td>
<td>0</td>
<td>2</td>
<td>5</td>
</tr>
<tr>
<td>SB2</td>
<td>0</td>
<td>0</td>
<td>2</td>
<td>10</td>
</tr>
<tr>
<td>SB3</td>
<td>0</td>
<td>0</td>
<td>1</td>
<td>5</td>
</tr>
</tbody>
</table>

VoiceCon will engineer its WAN trunk circuits to support compressed voice traffic (G.729A algorithm voice codecs) among all IPTS network facilities inter-facility voice traffic. **Any additional PSTN trunk circuits required to support local survivability requirements must be identified.** Necessary common equipment must be included in the system configuration and pricing proposals and identified as such.

**Vendor Response Requirements**

Confirm that the proposed IPTS network solution satisfies the stated trunk circuit requirements; support of centralized long distance trunk resources at the VoiceCon HQ facility for the SB facilities; and automatic alternate routing of calls among all VoiceCon facilities across the WAN and PSTN.

Siemens Response: The HiPath 8000 solution as designed provides for all trunking listed in table 5.

**2.3.1 ISDN PRI Services**

All installed VoiceCon T-1 trunk circuits must support ISDN PRI features and functions for both local and long distance exchange carrier transmission services.

**Vendor Response Requirements**

Confirm that the proposed T-1 trunk circuit interfaces support ISDN PRI capabilities.

Siemens Response: The gateways proposed support the ISDN PRI capabilities.
2.4 IPTS Network Growth Requirements

VoiceCon anticipates that station capacity requirements at the HQ and RO facilities will increase approximately 50% for the expected installed life of the proposed IPTS network solution. Port capacity growth requirements at the SB1 and SB2 facilities are anticipated to increase by about 20%; no growth is anticipated at the SB3 facility.

Vendor Response Requirement:
Confirm that the proposed IPTS solution can satisfy VoiceCon station port growth requirements and associated trunk growth requirements at its HQ, RO, SB1 and SB2 facilities without replacing any hardware equipment at time of initial system installation and cutover. Hardware additions are permissible to support incremental port interface requirements.

Siemens Response: The HiPath 8000 has a capacity of 100,000 users. The solution as designed can be increased to meet the above requirement without any hardware equipment.
3.0 Port Interface and Traffic Handling Requirements

The proposed IPTS network solution must be able to support a mix of TDM/PCM and IP ports. For traffic design engineering calculations assume the following traffic requirements:

1. The average busy hour traffic for IP desktop station users will be rated at 10 CCS @ P.01. Assume a traffic mix pattern of 30% intra-network calls, 15% outgoing local trunk calls, 25% outgoing long distance trunk calls, and 30% incoming DID trunk calls.
2. Analog telephone station busy hour traffic will be rated at 3 CCS @ P.01. Assume a traffic mix pattern of 70% inter-network calls and 30% outgoing local trunk calls. All analog telephone station calls will be subject to toll restrictions.
3. Assume that busy hour traffic is rated at 36 CCS @ P.01 for each of the following port types: all PSTN and WAN trunk circuits; attendant consoles; modems; audioconferencing units; facsimile terminals; voice mail ports.

Vendor Response Requirement:
The proposed system must design and engineer their system to support the above traffic assumptions. Confirm you have satisfied this requirement.

Siemens Response: Siemens confirms that the system has been designed to these specifications.

3.1 Circuit Switched Network Design

The proposed IPTS solution must support a variety of peripheral ports and switched connections. Although it is not required to support traditional digital voice terminal equipment, the IPTS must support analog communications devices. Switched connections involving non-IP ports may be handled using a circuit switched network, media gateways/Ethernet switches, or a combination of both methods.

Vendor Response Requirement:
If the proposed IPTS network solution includes integrated circuit-switched hardware equipment, then briefly describe the characteristics of the offering. Include, at minimum, the following information: hardware cabinet description; CCS @ P.01 rating; center stage switch and local TDM bus time slot/talk slot capacities; interswitch link capacities; all redundant design elements and level of redundancy.

Siemens Response: The HiPath 8000 is a soft-switch and does not use integrated circuit-switched hardware.
### 3.2 Peer-to-Peer Communications for IP Station to IP Station Calls

All two-party voice calls between IP desktop stations located at VoiceCon facilities must be handled exclusively over the LAN/WAN infrastructure without any circuit switched connections. This is a **Mandatory** requirement.

**Vendor Response Requirement:**
Confirm that your proposed system satisfies this **Mandatory** requirement.

Siemens Response: The HiPath 8000 is a SIP overlay system and uses peer-to-peer communications for all IP station to station calling.

#### 3.2.1 IP Station Discovery

How do IP communications devices learn about their voice VLAN, including IP addresses, default gateways, call controller, TFTP server, QoS settings, VLANs, and other parameters. Does the proposed system solution employ proprietary protocols for IP communications devices to learn their voice VLAN or is an industry standard, such as Dynamic Host Control Protocol (DHCP) used?

Siemens Response: The HiPath 8000 users DHCP for all IP end-points. The DLS (Device Licensing Server) provides for mass provisioning of IP end-points. The default gateways, call controllers etc. are set in the database of the HiPath 8000.

#### 3.2.2 IP Station Power over Ethernet (PoE)

VoiceCon requires that the power option to support IP telephones conform to IEEE 802.3af Power over Ethernet (PoE) standards.

**Vendor Response Requirement:**
Confirm that the proposed IPTS solution supports the IEEE 802.3af specification for in-line of IP telephone equipment. Describe current, future and retrospective compatibility of all proposed equipment. If 802.3af is not supported, identify the PoE implementation being proposed.

Siemens Response: The HiPath 8000 and desk top’s conform to the IEEE 802.3af PoE standards.

#### 3.2.3 IP Station QoS
Vendor Response Requirement:
Describe the proposed IPTS solution’s capabilities to provide Layer 2 and Layer 3 QoS to IP stations to ensuring end-to-end quality of service. Include in the response what industry standards are deployed.

Siemens Response: Siemens Complies. The HiPath 8000 QoS Manager uses information from network components to monitor and report QoS for both Layer 2 and Layer 3. This allows for the creation of QoS reports.

3.3 Multi-Party Conference Calls
The proposed system must be able to support six party add-on conference calls among IPTS stations and off-network stations. The system must also support a minimum of three (3) off-network stations per multi-party conference call when required. The HQ IPTS must support a minimum of 20 simultaneous multi-party add-on conference calls (up to six parties per conference) and the RO IPTS a minimum of 10 simultaneous multi-party add-on conference calls (up to six parties per conference)

Vendor Response Requirement:
Briefly explain how multi-party add-on conference calls are handled if: 1) All parties are on-network IP stations; 2) There is a mix of on-network IP and off-network stations.

The explanation should identify any and all hardware and software requirements necessary to support multi-party add-on conference call requirements. Specify if peripheral hardware equipment, e.g., conference bridge servers, is required.

Siemens Response: The HiPath 8000 supports multiple party conferencing up to 48 parties. The system will support either 48 internal or external add on numbers. Each station, depending on class of service, has the ability to use this feature. A Media Server is required to access this feature and is included in the design of this system. To access this feature the user simply uses the conference feature key on the telephone. The feature can also be access via a feature access code from phones that do not have a conference feature key. While on a call the user simply presses the conference key. The secondary party is placed on hold and the user then dials the third, fourth, etc., party. The display on the phone indicates that the user is added and that the conference is in process.

3.4 VoIP Overflow Traffic
If available WAN circuits connecting the HQ, RO and all SB facilities are busy, call admission control levels are reached, or QoS levels are not satisfied on-
network voice traffic must be able to automatically overflow to PSTN trunk circuits.

**Vendor Response Requirement:**
Confirm that your proposed communications system supports overflow of voice traffic across VoiceCon locations if WAN links are not available or conditions are not acceptable. Also indicate if overflow traffic cab revert back to the WAN if conditions permit.

Siemens Response: The HiPath 8000 supports PSTN overflow if the WAN circuits are all busy.

### 3.5.0 Port Interface Circuit Cards

For each of the following port types, provide a brief description of the proposed port interface circuit card(s) and/or media gateway equipment included with the proposed IPTS to support analog, digital, and IP ports. Include in the descriptions below the number of port interface terminations for each port circuit card, and the number of available gateway channels for each media gateway unit.

### 3.5.1 IP Telephones (desktop instrument and PC client softphones, including Attendant Console Position) & IP Audioconferencing Units

**Vendor Response Requirement:**
Provide a brief description how all IP telephone types are logically and physically supported by the common control call telephony server. If direct call control signaling via the Ethernet LAN/WAN is not supported identify all intermediary carrier, signaling interface and/or media gateway equipment that is required.

Siemens Response: The IP telephones are directly supported by the HiPath 8000. The HiPath 8000 uses a back-to-back user agent to support all IP phones. When a call is requested the phones signals the HiPath 8000 and then the HiPath 8000 signals the called phone or gateway. When an acceptance is received by the HiPath 8000 the call is connected via the RTP stream. The HiPath 8000 continues to monitor the call connection state.

### 3.5.2 Analog telephones

**Vendor Response Requirement:**
Provide a brief description how analog telephones are logically and physically supported by the common control call telephony server, identifying all intermediary hardware elements necessary for control signaling transmission. Specify the number of circuit terminations per circuit board/module/media
Siemens Response: The HiPath 8000 supports analog telephones via a Mediatrix analog gateway. These gateways support either 2, 4 or 24 analog devices. As with the IP telephones the HiPath 8000 uses a back-to-back user agent. To establish a call the user of the analog phone goes off hook and this sends a request to the Mediatrix gateway. The gateway requests the HiPath 8000 set up the call and the HiPath 8000 establishes the call to called phone or gateway.

3.5.3 Facsimile terminal

Vendor Response Requirement:
Provide a brief description how facsimile terminals are logically and physically supported by the common control call telephony server, identifying all intermediary hardware elements necessary for control signaling transmission. Specify the number of circuit terminations per circuit board/module/media gateway.

Siemens Response: See 3.5.2
3.5.4. Modem
Vendor Response Requirement:
Provide a brief description how facsimile terminals are logically and physically supported by the common control call telephony server, identifying all intermediary hardware elements necessary for control signaling transmission. Specify the number of circuit terminations per circuit board/module/media gateway.

Siemens Response: See 3.5.2

3.5.5 Power Failure Transfer Station (PFTS)
Vendor Response Requirement:
Provide a brief description how analog telephone instrument Power Failure Transfer Stations (PFTSs) are logically and physically supported by the common control call telephony server, identifying all intermediary hardware elements necessary for control signaling transmission. Specify the number of circuit terminations per circuit board/module/media gateway.

Siemens Response: Siemens provides this via standard PFTS transfer devices at the local offices. The PFTS transfer device is located on the trunk side of the provided gateway and monitors the power status of the gateway. If the gateway should lose power the the PFTS transfer device switches the assigned trunks to the appropriate station.

3.5.6 GS/LS CO Trunk
Vendor Response Requirement:
Provide a brief description how GS/LS CO trunk circuits are logically and physically supported by the common control call telephony server, identifying all intermediary hardware elements necessary for control signaling transmission. Specify the number of circuit terminations per circuit board/module/media gateway.

Siemens Response: The HiPath 8000 supports GS/LS Co trunk circuits via the Mediatrix FXO gateway. The maximum number of circuits per gateway is four.

3.2.7 DS1/T-1 Carrier Interface Trunk
Vendor Response Requirement:
Provide a brief description how DS1-based T-1 carrier trunk circuits are logically and physically supported by the common control call telephony server, identifying all intermediary hardware elements necessary for control signaling transmission. Specify the number of circuit terminations per circuit
board/module/media gateway.

Siemens Response: The Siemens RG 8700 gateway supports the DS1/T1 carrier interface trunks. The RG 8700 can provide either 2, 8 or 16 T1 or E1 circuits to the HiPath 8000. As with the stations the user wishing to make a call sends a request to the HiPath 8000. The HiPath 8000 requests the trunk from the RG8700 gateway. When the gateway has acknowledged the connection is established and the call is made.

3.2.8 Other Trunk Interfaces

VoiceCon may need at some future time additional analog trunk interfaces, specifically Auxiliary, FX, and E&M Tie Line.

Vendor Response Requirement:
Provide a brief description of how additional analog trunk interface requirements can be logically and physically supported by the common control call telephony server, identifying all intermediary hardware elements necessary for control signaling transmission. Specify the number of circuit terminations per circuit board/module/media gateway.

Siemens Response: Siemens complies. The Mediatrix gateways proposed in the RFP will support analog trunk interfaces including Auxiliary, FX and E&E Tie Lines. The HiPath 8000 supports the Mediatrix gateway as described in section 3.5.2. It should be noted that the use of E&M Tie Lines on the HiPath 8000 is very limited since all communications is done via the data network.
4.0.0. Voice Terminal Instruments

The proposed communications system must be able to support a mix of analog and IP communications devices. VoiceCon will provide its own analog telephone instruments, fax terminals, and modems.

4.1 Regulation Requirements

All single- and multi-line IP phones will be manufactured in accordance with Federal Communication Commission hearing aid compatibility technical standards contained in Section 68.316. and the Telecommunication Act of 1996.

Vendor Response Requirement:
Confirm the proposed telephone equipment satisfies these requirements

Siemens Response: Siemens Complies

4.2 Desktop IP Telephone Instruments

VoiceCon has a requirement for several types of desktop IP telephone instruments:

- Economy
- Administrative
- Professional
- Executive

4.2.1 Economy Desktop IP Telephone Instrument

The Entry model will be used in common areas. It should have, at minimum, the following design attributes and features/functions:

Comply with hold button exception on all models

- 12 key dial pad
- Single line appearance
- Hold button On screen – part of optiGuide like functionality. No fixed key
- G.711/G.729 voice codecs
- Auto Self Discovery/DHCP
- Echo Canceller
- IEEE 802.af POE support

Vendor Response Requirement:
Confirm that your proposed Economy model satisfies all of the stated requirements and provide a brief product description that includes an illustration/photograph (PPT format, only) of the instrument. Indicate in your
response any and all requirements not satisfied.

Siemens Response: Siemens is proposing the OpenStage Model 20 phone.

(IP & TDM variant)
Display
• Tiltable graphical display, monochrome (2 lines with 24 characters each)
Keys
• Keypad
• 7 fixed feature keys (partly equipped with red LEDs)
• Volume control
• 3-way navigation element (up-/down-/Enter-Key)
Acoustics
• Hands free talking (full duplex)
Others
• Wall mountable
IP specifics:
• CorNet IP & SIP
• Integrated 3-port Ethernet switch (10/100 Base-T)
• Codecs: G.711, G.729, G.722, iLBC for SIP
• 802.3af Power over LAN
• LAN activity LED
• Gigabit variant available

4.2.2 Administrative Desktop IP Telephone Instrument

The Administrative model will be used by station users who have executive management group call answering and coverage responsibilities. It should
have, at minimum the following design attributes and features/functions:

- 12 key dial pad
- Sixteen (16) programmable line/feature keys with soft label/status indicators 6 programmable keys with one being shift providing 10 keys. Could propose model 60 with 14 programmable keys or 80 with 16 keys but very expensive. Key module available to provide up to 22 additional programmable keys
- G711, G729 and wideband, e.g., G.722, voice codecs
- Auto Self Discovery/DHCP
- Echo Canceller
- QoS Support (802.1p/Q, DiffServ)
- Hold key On screen hold key available during call
- Last Number Redial key
- Release key
- Message Waiting/Call Ringing indicator(s)
- Full Duplex Speakerphone
- Speaker/Mute key
- Volume Control keys/slide
- High resolution, backlit, monochrome greyscale pixel-based, graphical display screen with four (4) associated context sensitive soft keys
- LDAP access No LDAP with 40 must be 60 or 80
- Stored Call Data (Last 10 numbers dialed/Last 10 incoming call numbers)
- Integrated Ethernet switch and two (2) RJ-45 connector interface ports for 10/100 Mbps connectivity Gigabit variant available
- Headset interface
- IEEE 802.af POE support

The Administrative model must also be capable of supporting optional add-on key modules if an additional 12 programmable line/feature keys with soft label/indicator status if required at some future time. comply

**Vendor Response Requirement:**
Confirm that your proposed Administrative model satisfies all of the stated requirements. Provide a brief product description that includes an illustration/photograph (PPT format, only) of the instrument. Indicate in your response any and all requirements not satisfied. State which required feature-specific keys are not available, but softkey feature access can be used as an alternative.

Siemens Response: Siemens is proposing the OpenStage Model 40 phone.
(IP & TDM variant)

Display
• Tilttable graphical display, monochrome, backlit (6 line user interface)

Keys
• Keypad
• 8 fixed feature keys (partly equipped with red LEDs)
• 6 free programmable keys with red LEDs; function-, speed dial- or line-keys
• Volume control
• 5-way navigation element (left-/right-/up-/down-/Enter-Key)

Acoustics
• Hands free talking (full duplex)

Interfaces
• Headset jack

Others
• Optical call alert
• Wall mountable
• Support of optional Key Module
• Support of Busy Lamp Field (CorNet IP and TDM)

IP specifics:
• CorNet IP & SIP
• Integrated 3-port Ethernet switch (10/100 Base-T)
• Codecs: G.711, G.729, G.722, iLBC for SIP
• 802.3af Power over LAN
• LAN activity LED

TDM specifics:
• USB slave for 1st party CTI
• Support of external phone- & analog adapter
• Gigabit variant available
4.2.3 Professional Desktop IP Telephone Instrument

The Professional model will be used by VoiceCon managers. It should have, at minimum the following design attributes and features/functions:

- 12 key dial pad
- Six (6) programmable line/feature keys with soft label/status indicators
- G711, G729 and wideband voice codecs
- Auto Self Discovery/DHCP
- Echo Canceller
- QoS Support (802.1p/Q, DiffServ)
- Hold key On screen during call
- Last Number Redial key
- Release key
- Message Waiting/Call Ringing indicator(s)
- Full Duplex Speakerphone
- Speaker/Mute key
- Volume Control keys/slide Slider
- High resolution, backlit, monochrome greyscale pixel-based, graphical display screen with four (4) associated context sensitive soft keys Color with 320x240 pixels
- LDAP access
- Stored Call Data (Last 10 numbers dialed/Last 10 incoming call numbers)
- Integrated Ethernet switch and two (2) RJ-45 connector interface ports for 10/100 Mbps connectivity Gigabit variant available
- Bluetooth interface for wireless headset
- USB interface
- IEEE 802.af POE support

The Professional model must also be capable of supporting the following integrated feature/functions if required at some future time:

- Gigabit (10/100/1000 Mbps) Ethernet connectivity
- Embedded Web-browser applications

Vendor Response Requirement:
Confirm that your proposed Professional model satisfies the stated requirements and provide a brief product description that includes an illustration or photograph (PPT format, only) of the instrument. Indicate in your response any and all requirements not satisfied. State which required feature-specific keys are not available, but softkey feature access can be used as an alternative.

Siemens Response: Siemens is proposing the OpenStage Model 60
telephones.

(IP & TDM variant)

Display
- Tiltable graphical color TFT display, 320x240 (QVGA) pixel, backlit

Keys
- Keypad
- 6 fixed feature keys (partly equipped with blue LEDs)
- 8 free programmable keys with blue LEDs (sensor keys); function-, speed dial- or line-keys
- 6 sensor mode keys with blue/white LEDs (to start applications)
- Capacitive volume slider with blue/white LEDs
- 'OpenStagewheel' (sensorial navigation element)

Acoustics
- Hands free talking (full duplex)
- High quality ringer tones
- Voice dialing

Interfaces
- Headset jack
- Bluetooth
- USB master
- USB memory stick data storage
- Connection of USB-WLAN dongle (for IP variant)
- Acoustic line in/out & recorder interface (via USB acoustic adapter)

Others
- Optical call alert
- Application support (see chapter 2.3)
- Support of optional Key Module
- Support of external OpenStage Keyboard
- Supported by “Phone Manager SW”

IP specifics:
• CorNet IP & SIP
• Integrated 3-port Ethernet switch (10/100 Base-T)
• Codecs: G.711, G.729, G.722, iLBC for SIP
• 802.3af Power over LAN
• LAN activity LED

TDM specifics:
• USB slave for 1st party CTI
• Support of external phone- & analog adapter

Gigabit variant available:
• Provides the same feature set as Workpoint 3 IP plus Gigabit Ethernet switch instead of a 10/100 MBit/s switch.

4.2.4 Executive Desktop IP Telephone Instrument

The Professional model will be used by VoiceCon’s executive management team. It should have, at minimum the following design attributes and features/functions:

- 12 key dial pad
- Twelve (12) programmable line/feature keys with soft label/status indicators
- G711, G729 and wideband voice codecs
- Auto Self Discovery/DHCP
- Echo Canceller
- QoS Support (802.1p/Q, DiffServ)
- Hold key
- Last Number Redial key
- Release key
- Message Waiting/Call Ringing indicator(s)
- Full Duplex Speakerphone
- Speaker/Mute key
- Volume Control keys/slide Slider
- High resolution, backlit, color pixel-based, graphical display screen with four (4) associated context sensitive soft keys VGA color 640x480 pixels
- LDAP access
- Stored Call Data (Last 10 numbers dialed/Last 10 incoming call numbers)
- Integrated Ethernet switch and two (2) RJ-45 connector interface ports; 10/100 Mbps connectivity Gigabit variant available
- Headset interface (Bluetooth is also acceptable)
- IEEE 802.af POE support

The Professional model must also be capable of supporting the following integrated feature/functions if required at some future time:

- Gigabit Ethernet connection
- Embedded Web-browser applications
Vendor Response Requirement:
Confirm that your proposed Executive model satisfies the stated requirements and provide a brief product description that includes an illustration or photograph (PPT format, only) of the instrument. Indicate in your response any and all requirements not satisfied. State which required feature-specific keys are not available, but softkey feature access can be used as an alternative.

Siemens Response: Siemens is proposing the OpenStage Model 80 telephone.

(IP & TDM variant)
Display
• Tilttable graphical color TFT display, 640x480 (VGA) pixel (IP variant), backlit
Keys
• Keypad
• 6 fixed feature keys (partly equipped with blue LEDs)
• 9 free programmable keys with blue LEDs (sensor keys); function-, speed dial- or line-keys
• 6 sensor mode keys with blue/white LEDs (to start applications)
• Capacitive volume slider with blue/white LEDs
• ‘OpenStagewheel’ (sensorial navigation element)
Acoustics
• Hands free talking (full duplex)
• High quality ringer tones
• Voice dialing

Interfaces
• Headset jack
• Bluetooth
• USB master
• USB memory stick data storage
• Connection of USB-WLAN dongle (for IP variant)
• Acoustic line in/out & recorder interface (via USB acoustic adapter)

Others
• Optical call alert
• Application support (see chapter 2.3)
• Support of optional Key Module
• Support of external OpenStage Keyboard
• Supported by “Phone Manager SW”

IP specifics:
• CorNet IP & SIP
• Integrated 3-port Ethernet switch (Gigabit variant available)
• Codecs: G.711, G.729, G.722, iLBC for SIP
• 802.3af Power over LAN
• LAN activity LED

TDM specifics:
• 320x240 (QVGA) pixel display with 8 free programmable keys
• USB slave for 1st party CTI
• Support of external phone- & analog adapter

4.2.5 Desktop IP Telephone Instrument Web-browser Functionality

Vendor Response Requirement:
Provide a brief description of embedded Web-browser functionality for the proposed Professional IP desktop telephone instrument model. Include the following information in your response: browser protocol (HTML, XML, WAP, Java, LDAP, Stimulus, other); station user interaction (touchscreen and/or keypad control cursor control; ability to place calls during active screen applications; screen saver option; standard and optional applications (visual mailbox; personal directory and calendar; web page access and display; visual alerts; audio alerts; et al).

OpenStage model 60 & 80 provide support the protocols below. All futures on applications support are scheduled for March 2007. Both models provide access to features/functions on the screen via the touch wheel at the center of the phone plus the phone supports a USB keyboard for easy and quick data input. Touch sensitive keys with LED for mail box, applications access, phonebook access, telephony, call log and help screen. The phone supports personalized ringers, caller ID with pictures, personalized screen savers and company logos, etc. At the top of the display there is a blue optical call alert.

<table>
<thead>
<tr>
<th>Application type</th>
<th>Examples</th>
</tr>
</thead>
</table>

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### 4.2.5 Desktop Instrument Options and Add-on Modules

**Vendor Response Requirement:**
Provide a brief description of available hardware/software options and/or add-on modules available with the proposed Economy, Administrative,
Professional, and Executive models. Options/modules should include key modules, display modules, Gigabit Ethernet connectors, and et al. necessary to satisfy the above telephone model requirements. Indicate the specific models that support the listed option/module.

External modules and adapters are scheduled to GA in March 2007 including Gigabit Ethernet variants of each model for SIP. HFA is schedules to GA in May 2007 for models 60 and 80 and July for HFA models 40 and 20.

The following accessories can be connected to the OpenStage workpoints:

**OpenStage Key Module**
- 12 additional free programmable keys with LEDs; function-, speed dial- or line-keys
- 1 Shift key with LED
- Large graphical display for key labeling
- For models 40, 60 & 80

**OpenStage Busy Lamp Field**
- Key extension with 90 free programmable keys with LEDs
- For CorNet IP & TDM, Workpoint 2

**OpenStage Speaker/Microphone Unit**
- External speaker & microphone in one housing
- Mute button with LED
- Volume control
- For models 60 & 80

**OpenStage Keyboard**
- External keyboard for text editing
- German (QWERTZ) and international (QWERTY) variant
- Fits to OpenStage design
- For models 60 & 80

**USB Acoustic Adapter**
- Allows connection of:
  - Speaker/Microphone Unit
  - Recorder / PC
  - USB Hub
- For models 60 & 80

**Phone- & Analog Adapter**
- Allows either the connection of:
  - 2nd OpenStage workpoint with UP0/E or
  - a/b device (Telephone / Fax)
- For all TDM workpoints

**Wall mount kit**
- For models 20 & 40
Key Module
40, 60, 80
12 free programmable keys with graphical display (multi layers)

BLF 40 Lamp
90 free programmable keys with graphical display (multi layers)

Acoustic Unit
External speaker & microphone in one housing
Mute button with LED
Volume control

Keyboard
QWERTZ & QWERTY variant
Fits to OpenStage design
Will be available for...

USB Extension
With class D amplifier to connect
In
4.2.6  **SIP Compatibility**

It is desirable, but not required, that the proposed desktop IP telephone instruments conform to current SIP standards and specifications at time of installation and system cutover. If any or all the proposed instrument models do not support natively embedded SIP capabilities, then it is acceptable that a firmware download upgrade be available when requested.

**Vendor Response Requirement**

Indicate which of the proposed telephone models are native SIP or can be reprogrammed via a SIP firmware download when requested by VoiceCon. Identify any proposed models that cannot currently be programmed for SIP support based on current commercial availability standards.

All models support SIP standard

Siemens Response:  Siemens Complies.

4.3  **PC Client Softphone**

A PC client softphone will be used by station users and attendant Operators as their primary desktop voice terminal. The PC client softphone application should conform to SIP standards and specifications.
Vendor Response Requirement
Confirm that the proposed softphone solution satisfies the stated SIP requirement.

Siemens Response: Siemens Complies

4.3.1 Desktop Station User Application

The proposed PC client softphone solution must be able to support a minimum of six programmable line appearances, integrated system and personal directories with search/dial-by-name capabilities, and functions comparable to the proposed Professional model. The softphone solution must also be able to support a peripheral headset.

Vendor Response Requirement
Confirm that the proposed softphone solution satisfies the stated requirements and provide a brief product description that includes a illustration/photograph (PPT format, only) that depicts the look and feel of an active call screen display.

Siemens Response: Siemens Complies
4.3.1.1 Teleworker Station User Application

The proposed PC client softphone solution may also be used by some station users as a teleworker voice terminal outside the VoiceCon facility environment.

Vendor Response Requirement
Confirm that the proposed softphone solution can be used as a teleworker voice terminal option. Indicate in your response if any optional hardware/software requirements are required to support teleworker mode Operations for deployment in a home, hotel, or office environment. If customer network requires software such as Nortel Connectivity for remote access would be the only exception.

Siemens Response: The Siemens OptiClient 130S can be used for the remote worker. Connection can be made via the VoiceCon VPN. This will provide full features and functionality. If a VPN connection is not used then Siemens recommends a Session Boarder Controller at the HiPath 8000 to prevent unauthorized access.

4.3.2 Soft Attendant Console

Attendant Operator console requirements are to be satisfied using a PC client softphone application. The attendant console application should include several distinct display fields, such as: incoming call queue and active caller information; release loop keys; feature/function keys; direct station selection (contact directory)/ busy lamp field; trunk groups; minor/major alarms; and messaging. GUI capabilities must support drag & click Operations.
At minimum the following information and data must be available in the softphone screen display: # Calls in queue; Call appearance status; Calling/called party number/name; Trunk ID; COS/COR; # Calls waiting; call coverage status; time/date, call duration; text messages; alarm notification

**Vendor Response Requirement**
Confirm the proposed softphone solution satisfies the stated requirements, and provide a brief description of the proposed softphone solution when programmed for attendant console operation. Include in the response a representative illustration or photograph (PPT format, only) that conveys the look and feel of an active call console display screen.

Siemens Response: The softphone that is used in conjunction with the console is the OptiClient 130 S. The HiPath 8000 console is based on HiPath ProCenter call center application. Both the OptiClient 130S and the HiPath ProCenter desk top are used in the console operation. The diagram below represents the HiPath 8000 Console Desk Top.

- One-click access to all telephony controls via softphone
- Alternatively, telephone keys can also be used
Indicates the device status of every user on a HiPath 8000
One-click collaboration with transfer to user extension or voicemail
Enhanced LDAP directory search
Supports recall handling to avoid unsuccessful transfers
Broadcaster “ticker tape” with real-time performance statistics for Attendant Groups
Overflow routing rules as well as additional attendants
Full contact center functionality for handling multimedia contacts

Manager Application Allows Managing and Monitoring Console Groups:
- Real-time and cumulative views with trend analysis, alerts and notifications
- Detailed historical reports for console group performance
5.0 Call Processing Features

The proposed communications system should have a robust list of call processing features supporting station user, attendant, and system operations.

5.1 Station User Features

It is required that the proposed communications system support the following list of station user features. Definitions for most listed features may be found in *PBX Systems for IP Telephony* (2002), written by Allan Sulkin and published by McGraw-Hill Professional.

Table 9 Station User Features

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<th>STATION USER FEATURES</th>
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<tr>
<td>EXECUTIVE ACCESS OVERRIDE</td>
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<tr>
<td>EXECUTIVE BUSY OVERRIDE</td>
</tr>
</tbody>
</table>
Vendor Response Requirement
Confirm that the proposed communications system supports each of the
above listed station user features. Identify any and all features that are not
included as part of the standard call processing software generic package.
Identify any and all of the listed features that require additional hardware
and/or software, e.g., CTI application server, because they are not included as part of the standard generic software package.

Siemens Response: All of the features listed in Table 9 are supports by the proposed OpenStand Phones. Some of the functions require additional applications on the HiPath 8000. Those features are listed below with the application that is needed.

**DISCRETE CALL OBSERVING** – HiPath ProCenter or other Call Center Application
**SILENT MONITORING** – HiPath ProCenter or other Call Center Application
**SUPERVISOR/ASSISTANT CALLING** – HiPath ProCenter or other Call Center Application
**SUPERVISOR/ASSISTANT SPEED DIAL** – HiPath ProCenter or other Call Center Application

### 5.1.1 Additional Station User Features

**Vendor Response Requirement**
Provide a listing of proposed standard generic software station user features that are not included in **Table 9** that VoiceCon may find of use and benefit.

Siemens Response; Some of the features of the HiPath 8000 and the OpenStage phones that are in addition to the listed features are.

Anonymous Call Rejection
Boss/Secretary Feature
Bluetooth support
Business Groups
**Caller Identity Delivery Suppression**
**Calling Number Identity Suppression**
**Multi-time zone support**
**Multi-tenant support**
Select Call Rejection
Serial Ringing
**Teleworking**
Voice Dialing

### 5.2 Attendant Operator Features
It is required that the proposed communications system support the following list of attendant operator features. Definitions for most listed features may be found in *PBX Systems for IP Telephony* (2002), written by Allan Sulkin and published by McGraw-Hill Professional.

**Table 10 Attendant Operator Features**

<table>
<thead>
<tr>
<th>ATTENDANT OPERATOR FEATURES</th>
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</thead>
<tbody>
<tr>
<td>AUTO-MANUAL SPLITTING</td>
</tr>
<tr>
<td>AUTO-START/DON'T SPLIT</td>
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<tr>
<td>BACK-UP ALERTING</td>
</tr>
<tr>
<td>BUSY VERIFICATION OF TERMINALS/TRUNKS</td>
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<tr>
<td>CALL WAITING</td>
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<tr>
<td>CAMP-ON</td>
</tr>
<tr>
<td>CONFERENCE</td>
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<tr>
<td>CONTROL OF TRUNK GROUP ACCESS</td>
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<tr>
<td>DELAY ANNOUNCEMENT</td>
</tr>
<tr>
<td>DIRECT STATION SELECTION w/BLF</td>
</tr>
<tr>
<td>DIRECT TRUNK GROUP SELECTION</td>
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<tr>
<td>DISPLAY</td>
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<tr>
<td>INTERCEPT TREATMENT</td>
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<tr>
<td>INTERPOSITION CALL &amp; TRANSFER</td>
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<tr>
<td>INTRUSION (BARGE-IN)</td>
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<tr>
<td>OVERFLOW</td>
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<tr>
<td>OVERRIDE OF DIVERSION FEATURES</td>
</tr>
<tr>
<td>PAGING/CODE CALL ACCESS</td>
</tr>
<tr>
<td>PRIORITY QUEUE</td>
</tr>
<tr>
<td>RECALL</td>
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<tr>
<td>RELEASE LOOP OPERATION</td>
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<tr>
<td>SERIAL OPERATION</td>
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<tr>
<td>STRAIGHT FORWARD OUTWARD COMPLETION</td>
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<tr>
<td>THROUGH DIALING</td>
</tr>
<tr>
<td>TRUNK-TO-TRUNK TRANSFER</td>
</tr>
<tr>
<td>TRUNK GROUP BUSY/WARNING INDICATOR</td>
</tr>
<tr>
<td>TRUNK ID</td>
</tr>
</tbody>
</table>

**Vendor Response Requirement**

Confirm that the proposed communications system supports each of the above listed attendant operator features. Identify any and all features that are not included as part of the proposed standard generic software feature package. Identify any and all features that require additional hardware and/or software, e.g., CTI application server, not standard with the proposed system model(s).

Siemens Response: The HiPath 8000 Console does not support the following functions in the table above.
AUTO-MANUAL SPLITTING

AUTO-START/DON'T SPLIT

CONTROL OF TRUNK GROUP ACCESS – The HiPath 8000 uses gateway selection rather than trunk group access and therefore this feature is not included in the HiPath 8000 Console operation.

DIRECT STATION SELECTION w/BLF – The HiPath 8000 Console does support direct station signaling via an integrated LDAP directory and a presence indication therefore use to BLF is no longer necessary.

DIRECT TRUNK GROUP SELECTION – The HiPath 8000 routing provides for direct access to the appropriate trunking gateway. This ensures that the call will go out the best available route. Operator selection of trunk groups is not available in the HiPath 8000 due to this feature.

INTERPOSITION CALL & TRANSFER

INTRUSION (BARGE-IN)

OVERRIDE OF DIVERSION FEATURES

PAGING/CODE CALL ACCESS – The HiPath 8000 does allow for paging access from the console with an optional paging system.

RELEASE LOOP OPERATION – The HiPath 8000 is a SIP based communications systems and circuits are not used in the traditional manner. The calls are sent over the data network as packets and therefore release loop operation is not used in the HiPath 8000.

TRUNK GROUP BUSY/WARNING INDICATOR – As with direct trunk group selection this function is not used in the HiPath 8000.

5.2.1 Additional Attendant Operator Features

Vendor Response Requirement
Provide a listing of proposed standard generic software attendant operator features that are not included in Table 10 that VoiceCon may find of use and benefit.

Siemens Response:
Attendant Desktop Features

The HiPath ProCenter Attendant Console offers an intuitive user interface, reducing training time for attendants. Flexible tearoff-and-park toolbars allow users to adjust the desktop display to their preferred working style.

Receiving and Handling Calls

Synchronized with every incoming call, the attendant receives a screen-pop window with information such as the caller's number, the called number, a call description and information on the caller's wait time before being connected, providing insight into the caller's experience. A fully integrated LDAP directory search allows attendants to quickly find the required contact information. For subscribers on the HiPath 8000, the LDAP directory search window will also display line states, such as "idle" or "busy". This enables the attendant to determine whether the caller should be transferred to the extension, or to voicemail, if the line is busy or the phone is ringing. Attendants can transfer a calls, initiate consultations or conference in other participants with just a click of the mouse.

Recall Handling

When the targeted person does not answer the phone, the call is automatically redirected back to an available attendant. This attendant receives a screen-pop window clearly indicating the recall status in call description and queue name. With the extended call history provided in the recall screen-pop window, attendants can track the call's history prior to arriving at their desk.

Productivity Tools for Attendants

The Attendant Console Desktop offers users productivity tools that help to optimizing collaboration with other attendants, and provide constant feedback on the operational status of attendant groups, as well as the productivity of every attendant.

Attendants can use the Team List and Team Bar features to "see" the availability status of attendants in their own or other attendant groups, enabling them to transfer callers with the knowledge that they will not end up in queue again. The Broadcaster “ticker tape” integrated into the agent desktop can be used to display real-time operational statistics, as well as messages or alerts for the attendant group. Personal performance statistics give attendants insight into their productivity, and the contacts waiting indicator enables them to pace call handling according to current operational conditions.

Extended Attendant Desktop Features

As an option, the Attendant Console Desktop may be extended to comprise full multimedia functionality for handling email and live web contacts in addition to incoming voice calls. This will allow attendants to seamlessly interface with the contact center operations in your enterprise, whenever
Manager Desktop Features

To streamline the administration of users and attendant groups, HiPath ProCenter Enterprise provides a unified and easy-to-use interface for all management tasks. The Manager Desktop is a highly visual and easily customizable console organized into "work centers" dedicated to key management tasks:

Administer attendant desktop users and resources
Define overflow routing rules as well as additional attendants in a highly usable, visual interface
Define real-time monitoring views and historical reports to optimize console group management
Create streaming Broadcaster "ticker tape" content for attendant desktops

Administration Center
Administration Center is a convenient interface for the administration of attendant console users, user profiles, and user groups. Wrap-up Reason codes to document call handling activity are also configured here.

Design Center
Design Center gives attendant group managers a visual, workflow-style tool for defining overflow routing rules. Configurable and reusable components simplify creating and editing routing flows in a drag-and-drop interface. Strategies are automatically checked and validated for completeness as they are created. Design Center offers components such as: Time of day / day of week schedules to treat incoming calls according to the time of their arrival Source / destination routing decisions to determine treatment of incoming calls based on the number the caller dialed or the caller’s telephone number Performance level routing decisions to ensure incoming calls are handled promptly even during times of higher than usual incoming call volumes Data directed routing decisions that allow to automatically interface with external ODBC-compatible databases, as well as integrated LDAP directories

Report Center
The flexible interface of HiPath ProCenter Report Center makes tailoring specific reports or formats easy. Real-time and cumulative views with trend analysis, alerts and notifications enable attendant group managers to optimally monitor attendant group performance and incoming call volumes. Detailed historical reports for console group performance give managers insight into day-to-day operations and simplify planning.

Broadcast Center
Broadcast Center offers a fully integrated interface for defining rules-based streaming statistics in Broadcaster "ticker-tape" views for the attendant console desktop. Managers can configure rules-based thresholds for broadcaster views, to alert attendants visually of changes in the operational conditions. Easy to define distribution lists for broadcaster views ensure that the relevant data is sent to a selected group of recipients.

**Attendant Console Desktop Summary**

Attendant Console is an optional desktop of HiPath ProCenter Enterprise

- Full desktop telephony controls; plus click-to-dial speed dial, directory integration and contact log
- Call details screen pop window synchronized with each incoming call
- Unique multimedia presence management and collaboration tools
- One-click collaboration with transfer to user extension or voicemail
- Enhanced LDAP directory search
- Indication of device status of subscribers on HiPath 8000
- Recall handling support to avoid unsuccessful transfers
- Availability and call wrap-up reason codes
- Broadcaster “ticker tape” with real-time performance statistics for Attendant Groups
- Customizable launch pad with “tear off and park” toolbars
- Optional multimedia contact handling for E-mail and live Web interactions

**Manager Desktop Summary**

- User, attendant group and data source administration
- Visual interface for designing call handling and overflow rules
- Graphical real-time and historical monitoring and reporting, alerts and notifications
- Rules-based, streaming broadcast capabilities for attendant console desktops
- Telephony platform synchronization and related capabilities
5.3 System Features

It is required that the proposed communications system support the following list of system features. Definitions for most listed features may be found in *PBX Systems for IP Telephony* (2002), written by Allan Sulkin and published by McGraw-Hill Professional.

**Table 11 System Features**

<table>
<thead>
<tr>
<th>SYSTEM FEATURES</th>
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</thead>
<tbody>
<tr>
<td>ACCOUNT CODES</td>
</tr>
<tr>
<td>ADMINISTERED CONNECTIONS</td>
</tr>
<tr>
<td>ANSWER DETECTION</td>
</tr>
<tr>
<td>AUTHORIZATION CODES</td>
</tr>
<tr>
<td>AUTOMATED ATTENDANT</td>
</tr>
<tr>
<td>AUTOMATIC CALL DISTRIBUTION</td>
</tr>
<tr>
<td>AUTOMATIC ALTERNATE ROUTING</td>
</tr>
<tr>
<td>AUTOMATIC CAMP-ON</td>
</tr>
<tr>
<td>AUTOMATIC CIRCUIT ASSURANCE</td>
</tr>
<tr>
<td>AUTOMATIC NUMBER ID</td>
</tr>
<tr>
<td>AUTOMATIC RECALL</td>
</tr>
<tr>
<td>AUTOMATIC ROUTE SELECTION – BASIC</td>
</tr>
<tr>
<td>AUTOMATIC TRANSMISSION MEASUREMENT SYSTEM</td>
</tr>
<tr>
<td>CALL-BY-CALL SERVICE SELECTION</td>
</tr>
<tr>
<td>CALL DETAIL RECORDING</td>
</tr>
<tr>
<td>CALL LOG</td>
</tr>
<tr>
<td>CENTRALIZED ATTENDANT SERVICE</td>
</tr>
<tr>
<td>CLASSES OF RESTRICTION (SPECIFY #)</td>
</tr>
<tr>
<td>CLASSES OF SERVICE (SPECIFY #)</td>
</tr>
<tr>
<td>CODE CALLING ACCESS</td>
</tr>
<tr>
<td>CONTROLLED PRIVATE CALLS</td>
</tr>
<tr>
<td>DELAYED RINGING</td>
</tr>
<tr>
<td>DIAL PLAN</td>
</tr>
<tr>
<td>DIALED NUMBER ID SERVICE</td>
</tr>
<tr>
<td>DIRECT DEPARTMENT CALLING</td>
</tr>
<tr>
<td>DIRECT INWARD DIALING</td>
</tr>
<tr>
<td>DID CALL WAITING</td>
</tr>
<tr>
<td>DIRECT INWARD SYSTEM ACCESS</td>
</tr>
<tr>
<td>DIRECT INWARD TERMINATION</td>
</tr>
<tr>
<td>DIRECT OUTWARD DIALING</td>
</tr>
</tbody>
</table>
Siemens Response: The HiPath 8000 system supports all of the features in Table 11 with the exception of the features listed below. If those features are supported by another application the application is listed next to the
feature.

AUTOMATED ATTENDANT – Optional supported by Media Server
AUTOMATIC CALL DISTRIBUTION – HiPath ProCenter or other Call Center Application
DIRECT INWARD SYSTEM ACCESS – HiPath OpenScape
FORCED ENTRY ACCOUNT CODES – HiPath ComAssistant or other Click and Dial Software
INTEGRATED SYSTEM DIRECTORY – HiPath ComAssistant, HiPath ProCenter, HiPath OpenScape
6.0.0 Systems Management

The proposed communications system must be administered, monitored, and maintained through operations organized into five functional areas: Fault, Configuration, Accounting, Performance and Security. All of the systems and devices in your proposed solution should attempt to provide comprehensive operations in each area.

Operations for each area must be accessible through one interface regardless of the underlying system or device being managed. If a proxy server is used for intermediate operations, there must be at most one central database for each functional area. Systems or devices may be accessed individually if no proxy server is used.

EXCEPTION: Optional call center solutions may provide its own set of FCAPS management operations separate from the general enterprise communications solution.

Any supplied management applications must support decentralized access from any distributed PC client across the HQ LAN/WAN infrastructure and remote dial-up PC clients. It is also desirable for the applications to support a browser based user interface for intensive remote operations.

Any supplied management applications may integrate information from the five functional areas at the presentation level.

Vendor Response Requirement

Confirm and verify that each functional area required to manage the proposed IPTS network is supported by a single, centrally located proxy server or, alternatively, each system or device supports a single API for a given functional area. Provide a brief description of the proposed management system, including its major hardware and software components. Specify if the proposed systems management server and software is available as a bundled offering, only, or if VoiceCon is responsible for providing its own server hardware to operate the software. If third party technology is used, please indicate which components are managing your solution in a vendor agnostic fashion.

Siemens Response: The management of the proposed solution is architecturally aligned with the centralized approach of the HiPath 8000.

FCAPS functions:
HiPath 8000 Assistant:

The HiPath 8000 Assistant provides the graphical user interface for system administration of HiPath 8000 subscriber information, gateways and phones. It runs on the same platform as the HiPath 8000 itself i.e. it uses LINUX as the operating system. The HiPath 8000 Assistant shows a consolidated
status overview on the splash screen. It indicates with color coded icons whether there are any alarms prevalent on the system.

For call accounting HiPath 8000 provides traditional CDR records that can be collected and used for customized reports/evaluations.

HiPath Xpressions:
The basic configuration of the UM application is done through its own management application. Adding mailboxes to subscribers can be done through the same user interface that is used to add or delete HiPath 8000 subscribers. Xpressions runs on a Windows platform and provides a browser-based graphical user interface.

Please note that the HiPath 8000 Assistance and the HiPath Xpressions Manager can be managed by the HiPath Manager. The HiPath Manager is a single interface that allows the administrator to use a single terminal and a single GUI to interface with the various elements of the HiPath 8000 solution as proposed.

6.0.1 System/Port Capacity
Vendor Response Requirement
Identify the maximum number of independent IPTS communications systems that can be supported by the proposed systems management server, and the maximum number of user ports that can be passively and actively supported.

Siemens Response: The HiPath 8000 Assistant supports a maximum of 100,000 users with the Assistant running on a separate server.

6.0.2 Terminal Capacity
Vendor Response Requirement
Identify the maximum number of configurable and active PC client terminals that can be configured as part of the proposed management server system.

Siemens Response: The HiPath Assistant can support up to 100,000 active users regardless of the type of IP end point.

6.0.3 Support for Open Standards
The proposed management system should provide support for open protocols, such as LDAP and SNMP. The proposed management system should use open encoding schemes, such as XML and HTML.

Vendor Response Requirement
Briefly discuss the open standards included in your proposed management system that supports administration, operations and maintenance services. Indicate if any protocols or encoding schemes are de facto standards or are
being implemented publicly by other vendors.

Siemens Response: The HiPath 8000 Assistant uses the open SOAP interface of the HiPath 8000. This interface is publicly available to other vendors. The same is true for the SNMP interface that is provided to send alarm information to a surveillance function in the network. The Private MIB is made publicly available to customers and third party vendors.

HiPath User Management provides the integration of the subscriber administration on the graphical user interface level. It allows for integration of HiPath 8000 subscriber data and Xpressions subscriber data in an LDAP directory. This in turn can be integrated with any LDAP interface a customer may have in their environment.

6.0.3 Security Features
Unauthorized access to the communications system is a major concern. The ability to detect security problems is desirable beyond mechanisms to prevent security problems.

Vendor Response Requirement
Briefly describe the security features that are embedded in the proposed management system to prevent unauthorized access and operation. Specify if media encryption is used for command signaling transmissions. What, if any, Denial of Service (DoS) and user authentication mechanisms are supported for the systems management application?

Siemens Response: The HiPath 8000 Assistant provides the ability to create users for any kind administrator role a customer might need. So access rights can be defined for areas of the product for every administrator specifically. Access of these administrators is protected by user names and passwords.

In addition an audit trail is part of the product. It allows for an after-the-fact lookup which administrator has done what and when.

If there is a malicious attempt to break into the system by trying to find out the password HiPath 8000 reacts appropriately. A threshold can be defined after how many unsuccessful login attempts the system will send an SNMP alarm to the surveillance system. In addition the login account used will be locked out and the User Administrator will have to specifically unlock that account manually.

6.0.5 User Interface & Tools
The management system should be operated using by GUI tools, formatted screens, pull down menus, valid entry choices, templates, batch processing & transactions scheduling, and database import/export. In general you should
support a user interface set for each functional area: fault, Configuration, Performance and Security. The constituent users of each of these areas are distinct and your interface for each should optimize the experience for that constituent group. Management applications may integrate information from several management areas to enhance one functional area being managed.

Siemens Response: All management functions of the proposed solution have GUI tools and all other characteristics described above.

6.1.0 Administration Functions
The proposed systems management solution must support: station user moves, adds, and changes; trunk group definitions and individual trunk circuit programming; voice terminal parameters; call restriction assignments; class of service definitions and assignments; password resets; customer profile database; ARS routing tables; group definitions and assignments; first digit tables; dial plan; feature access codes; paging/code call zone assignments.

Vendor Response Requirement
Confirm the proposed systems management solution supports each of the listed administrative functions. Identify any functions not supported.

Siemens Response: The proposed solution contains all functions in either GUI or CLI format. Paging/code call zone assignment is not applicable
**Group Assignments**
The administration subsystem must support each of the following group definitions and assignments

- Abbreviated Dialing (System, Group, Enhanced)
- Hunt Groups
- Call Coverage Answer Groups
- Pickup Groups
- Intercom Groups
- Terminating Extension Groups
- Trunk Groups

**Vendor Response Requirement**
Confirm administration support for each of the listed group definitions. List any and all groups not supported by the administration subsystem.

Siemens Response: The proposed administration subsystem supports all of the above group configurations. We understand that Call Coverage Answer Groups are represented in the softswitch as Pickup Groups. Trunk Groups are configured in the softswitch as a portion of a gateway and are addressable as such.

**6.1.2 Facilities Performance Management & Reports**
The management system must be able to collect, analyze, and provide reports for a variety of system operations.

Siemens Response: The Basic Traffic tool provides a graphical and numerical representation of collected system data e.g. number of incoming calls, number of outgoing call, number of failed calls and BHCA.

**6.2.1 Basic Trunk Usage and Traffic**
Trunk traffic records should be kept for all inbound and outbound calls, identifying the trunk group and trunk channel, time and duration of call.

**Vendor Response Requirement**
Confirm that the proposed facilities management system satisfies this requirement.

Siemens Response: The HiPath Accounting Management manages the traffic records collected in the HiPath 8000. This includes the records of incoming
and outgoing calls routed through gateways. In a survivability situation the
records are generated in the gateway. The trunk group and channel, time
and duration are recorded in the traffic records

6.2.1.1 Individual Trunk Line Counters
Vendor Response Requirement
Confirm that individual trunk line counters measure and report: Number of
call attempts; Number of blocked trunk lines; Traffic intensity (Erlangs).

Siemens Response: The HiPath 8000 provides individual trunk line counters
in conjunction with the trunking gateways.

6.2.1.2 Outgoing Trunk Route Counters
Vendor Response Requirement
Confirm that outgoing trunk route counters measure and report: Number of
outgoing attempts; Number of successful calls overflowing to another route;
Number of lost calls due to blocking; Number of blocked trunks in
measurement; Traffic intensity (Erlangs).

Siemens Response: The HiPath 8000 provides trunk route counters in
conjunction with the trunking gateways.

6.2.1.3 Incoming Trunk Route Counters
Vendor Response Requirement
Confirm that incoming trunk route counters measure and report: Number of
incoming call attempts; Number of trunks in the measurement; Number of
blocked trunks in the measurement; Traffic intensity (Erlangs).

Siemens Response: The HiPath 8000 provides trunk route counters in
conjunction with the trunking gateways.

6.2.1.4 Both Way Trunk Route Counters
Vendor Response Requirement
Confirm that both way trunk route counters measure and report: Number of
incoming call attempts; Number of trunks in the measurement; Number of
blocked trunks in the measurement; Traffic intensity (Erlangs).

Siemens Response: The HiPath 8000 provides both way trunk route counters
in conjunction with the trunking gateways.

6.2.2 Attendant Consoles
Attendant counters should measure all attendants in the system, or
individual attendant positions. Record measurements include: number of
answered calls; number of calls initiated by attendant; accumulated handling time for all calls; accumulated handling time for recalls; accumulated handling time for calls initiated by attendant; accumulated total delay time for recalls; number of answered recalls; number of abandoned attendant recalls; accumulated waiting time for abandoned calls to an attendant; accumulated waiting time for abandoned recalls, and accumulated response time for all types of calls.

**Vendor Response Requirement**
Confirm that attendant counters measure and provide reports for each of the listed parameters. Identify attendant parameters which are not measured.

Siemens Response: The HiPath 8000 provides reports through the CDR system for all of the indicated parameters.

### 6.2.3 Stations

Station counters should measure individual stations or station group traffic statistics, including: number calls; number of stations in measurement; number of blocked stations in measurement; traffic rating (Erlangs).

**Vendor Response Requirement**
Confirm that station counters measure and provide reports for each of the listed parameters. Identify station parameters which are not measured.

Siemens Response: The HiPath 8000 provides very sophisticated CDRs. Typically in this solution customers will create such statistics using the Call Accounting tool. The product that Siemens provides is HiPath Accounting Management as part of the suite of HiPath Management applications.

Siemens Response: The HiPath 8000 provides

### 6.2.4 Traffic distribution

When applicable, traffic distribution across the internal switching network should be measured for each local TDM bus, traffic over each highway bus, and traffic across the center stage switch by each switch network interface link.

**Vendor Response Requirement**
Confirm that traffic distribution is measured and reported for each switch network element listed. Identify what is not measured and reported.
Siemens Response: The HiPath 8000 is a pure softswitch and does not use a TDM bus. Traffic distribution is reported by the HiPath 8000 via the CDR system.

6.2.5 Busy hour traffic analysis
Busy hour traffic analysis measurements for trunks, stations, and the internal switch network should be performed and reported for any one hour interval for any time of the day.

Vendor Response Requirement
Confirm busy hour traffic measurements for trunks, stations, and the internal switch network for any one hour interval for any time of the day.

Siemens Response: This HiPath 8000 can report on busy hour traffic analysis based on the desired customer parameters. Reports can be set for a one hour interval for any time of day.

6.2.6 Erlang Ratings
Erlang rating should be calculated and reported for individual trunk lines, each trunk group, and all trunk groups. CCS ratings should be calculated for individual stations or groups of stations.

Vendor Response Requirement
Confirm Erlang and CCS rating calculations and reporting for each listed item.

Siemens Response: Exception. The HiPath 8000 does not provide these calculations however optional 3 party software may be obtained to provide this information.

6.2.7 Processor Occupancy
System call processing performance is measured in terms of Busy Hour Calls (Attempts and Completions). The percent of maximum call processing capacity should be reported for programmed time intervals. Threshold reports should also be generated to monitor system load factors.

Vendor Response Requirement
Confirm measurement and reporting of processor occupancy and threshold levels

Siemens Response: The HiPath 8000 provides for BHCA of up to 235,000. Performance can be measured via optional software which is not included in this bid.
6.2.8 Threshold Alarms
For a variety of system hardware devices it should be possible to define a congestion threshold value, and measure generated alarms. Alarms are recorded in an Alarm Record Log. The types of devices that can be tracked include: tone receivers; DTMF senders and receivers; conference bridges; trunk routes; modem groups.

**Vendor Response Requirement**
Confirm recording and reporting of alarms for each listed item.

Siemens Response: The HiPath 8000 provides for threshold alarms. Note that the HiPath 8000 does not use tone receivers or DTMF senders and receivers.

6.2.9 Feature Usage
Feature usage counters for selected station features, e.g., call forward, call transfer, add-on conference, and attendant system features, e.g., recall, break-in, should be measured and reported for programmed intervals.

**Vendor Response Requirement**
Confirm recording and reporting of feature usage counters for both station and attendant operations.

Siemens Response: The HiPath 8000 provides feature usage counters.

6.2.10 VoIP Monitoring
The management system should collect and store data to track usage and performance data of IP gateway devices, IP phones, and VoIP intercom/trunk calls. VoIP information reports may include: tracking of IP gateway devices and calls that pass through each gateway; gateway congestion; assignment of services or routes to gateways; tracking of phone numbers dialed or originating off-site numbers; and IP gateway addresses.

**Vendor Response Requirement**
Briefly describe all VoIP monitoring information records and reports that are available. Specify if VoIP QoS parameters such as jitter, call delay/latency, and packet loss are tracked and reported, and if a system administrator can monitor VoIP calls in real-time for QoS observing? Indicate if any third party equipment is being proposed as part of your solution.
Siemens Response: The HiPath 8000 provides this capability in multiple ways. On the one hand the usage i.e. the load of certain parts of the network like gateways can be tracked by creating certain reports within the call accounting tool. On the other hand the performance information is provided by another one of the HiPath Management Applications which is called HiPath QoS Management. It creates reports based on information it gets from the network e.g. the endpoints about jitter, delay/latency and packet loss. The monitoring is near-real-time.

6.3 Optional Reports

Directory records may include each subscriber’s name along with a variety of phone numbers such as primary, published, listed, emergency, and alternate, as well as authorization code information, job title, employee number, current employment status and SSN.

Inventory records and management is used to administer any kind of inventory product part, including: PBX common equipment (cabinets, carriers, circuit cards); voice terminals and module options; jacks, and button maps. The reports allow administrators to accurately re-charge items. Inventory can be tracked by data such as user, system (PBX or other networks), jack, serial number, asset tags, trouble calls, recurring and non-recurring costs, and general ledger codes. The inventory management system may also include records containing the following data: purchase date, purchase order number, depreciation, lease dates, manufacturer and warranty information.

Cabling records keep track of all cable, wire pairs, distribution frames, wiring closets and all connections (including circuits) down to both the position and the pair level. Cable records include starting and ending locations, description, type and function. Individual cable lengths are maintained and automatically added, as is the decibel loss, for the entire path. Information can also be provided on the status of all cable runs, as well as the number of pairs it contains, the status of the pairs, and the type of service it provides.

Vendor Response Requirement

Identify and briefly describe your proposed management system’s Directory, Inventory, and Cabling reports, if available.

Siemens Response: The HiPath 8000 works together with the HiPath Management Application HiPath User Management. It is basically the subscriber directory database that allows storage of various kinds of information together with name, number, etc. It is not a cabling management tool per say. These tools are typically CAD based tools that
contain building and floor plans with locations for data closets and so on. HiPath User Management is more the subscriber LDAP directory application.

6.4.0 Call Detail Recording

Call Detail Record (CDR) data should be compiled for all successful incoming and outgoing trunk calls. Call record fields typically include the following:

- Date
- Time
- Call Duration
- Condition Code (categorizes information represented in the call record)
- Trunk Access Codes
- Dialed Number
- Calling Number
- Account Code
- Authorization Code
- Facility Restriction Level for Private Network Calls
- Transit Network Selection Code (ISDN access code to route calls to a specific inter-exchange carrier)
- ISDN Bearer Capability Class
- Call Bandwidth
- Operator System Access (ISDN access code to route calls to a specific network operator)
- Time in Queue
- Incoming Trunk ID
- Incoming Ring Interval Duration
- Outgoing Trunk ID

Vendor Response Requirement
VoiceCon will purchase its own third party call accounting and billing system. Identify all available CDR reports that can be generated for any or the entire call record field data listed above.

Siemens Response: Siemens Complies

6.5.0 Maintenance

System maintenance operations should, at minimum, support the following: Monitoring of processor status; Monitoring and testing of all port and service circuit packs; Monitoring and control of power units, fans, and environmental sensors; Monitoring of peripherals (voice terminals and trunk circuits);
Initiate emergency transfer and control to backup systems; Originate alarm information and activate alarms.

**Vendor Response Requirement**
Confirm support of each listed maintenance monitoring activity. Identify any activity not supported.

Siemens Response: The HiPath 8000 is built on a standard industry server and OS (LINUX). Both of which come with its specifically designed set of alarms. Also the application will trigger alarms for certain conditions within the softswitch application itself. Of course alarming for voice circuits and such are not applicable in this environment. However all network elements that are part of the solution (e.g. endpoints, gateways, etc.) have also the ability to send alarm information to a monitoring facility.

### 6.5.1 Alarm Conditions
There are usually several types of communications system alarm conditions: Major, Minor, and Warning.

**Vendor Response Requirement**
Briefly describe how your management system defines a Major, Minor, and Warning alarm.

Siemens Response: The HiPath 8000 provides alarms with all mentioned severities. The threshold condition is defined individually alarm by alarm. Documentation provides a description when a certain alarm issues a warning or a minor or major alarm state.

### 6.5.2 Maintenance Reports
**Vendor Response Requirement**
Identify any and all available maintenance alarm reports provided by your management system.

Siemens Response: Siemens Complies. The Administration user interface provides an overview of the current alarm status on it splash screen so the consolidated status of the softswitch system can be captured instantaneously. Alarm reports can be created as part of a network wide consolidated status report using HiPath 8000s capability to report the status into Enterprise network management applications like HP OpenView or MicroMuse.
6.5.3 Remote Maintenance
Vendor Response Requirement
Briefly describe the available options used to support remote maintenance operations for both customer access and for an outside maintenance service provider. Specify how the system alerts a remote service center when an alarm condition occurs, the trunk circuit requirements for alert transmissions, and security measures to prevent unauthorized access.

Siemens Response: The HiPath 8000 provides remote access capabilities. The interfaces are secure (SSH, SFTP and HTTPS). Remote administration can either be performed by using the CLI or the web-based user interfaces. The notification of alarms to a centralized NOC is done through standard SNMP traps.

6.6.0 Provisioning

All services should be provisioned in one step. Services should include station configuration, voice mailbox configuration, E-911 location, billing attributes, directory attributes, and mobile Email attributes (Blackberry) and the configuration of other end user applications.

For example, if your solution includes a zone paging application, the ability to assign a station to a zone and change the zone membership as a whole must be accessible through the configuration (provisioning) interface.

Templates must be supported to organize different settings across different systems according to organizational need. At a minimum, the voice station configuration and the associated voice mailbox must be provisioned in one step through one interface.

Your proposed provisioning application or interface must create a complete audit trail and must allow groups of changes to be scheduled for a future time. Further, the solution must support mass create, delete and modify functions to support bulk operations.

Vendor Response Requirement
Describe the provisioning workflow you recommend showing how each of your proposed solution components is utilized. List any functions above which are not available. List any systems or devices which are not now part of your provisioning interface and provide a roadmap statement of how you will treat this situation going forward.

Siemens Response: The HiPath 8000 provides with its HiPath User
Management an application that allows for the administration of subscribers together with its Xpressions voicemail configuration. Mobile email attributes are not supported at this time.

### 7.0.0 Integrated Messaging System

VoiceCon requires a HQ-based voice messaging system that must be fully integrated with the proposed IPTS network solution. VoiceCon also requires integration of the proposed voice messaging system with a Microsoft Exchange messaging system to provide “unified” messaging applications. The proposed voice messaging system solution must be centrally located at the VoiceCon HQ location, and be capable of supporting station users at all remote VoiceCon facilities (RO and SBs).

The voice mail system will also serve as an automated attendant position for select incoming trunk calls, and also as a secondary point of coverage as an automated attendant system for designated stations. All software and hardware necessary to interface with the existing telephone system will be provided under this bid.

The sizing requirements are:

<table>
<thead>
<tr>
<th>Installed/Equipped Capacity</th>
<th>Maximum Capacity</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of Users</td>
<td>2000</td>
</tr>
<tr>
<td>Number of Ports</td>
<td>64</td>
</tr>
<tr>
<td>Hours of Storage</td>
<td>1000</td>
</tr>
<tr>
<td></td>
<td>3,000</td>
</tr>
<tr>
<td></td>
<td>96</td>
</tr>
<tr>
<td></td>
<td>1200</td>
</tr>
</tbody>
</table>

Five (5) automated attendant ports are included in the requirements. A Grade of Service level of P.01 is required.

**Vendor Response Requirement**

Briefly describe the proposed integrated messaging solution, and provide details about the voice mail system architecture and its interconnection to the voice communications system and Microsoft Exchange system. Include processing system platform information in the discussion. Verify that the system being bid can comply with each of the proceeding requirements.

Siemens Response: Comply

HiPath Xpressions V5.0 is a Unified Messaging Service which integrates the following services: voice mail, fax mail, E-mail and Short Message Service (SMS).

HiPath Xpressions V5.0 is employed in modern enterprises for whom an
information flow, flexibility, and speed are decisive factors in their success.

Its modular, scalable client/server architecture means that HiPath Xpressions V5.0 can be adapted to fulfill enterprises’ communications requirements to the optimum. Customers can select services, users, DP integration, software-only solutions or certified complete systems according to their requirements. From the small-scale introductory variant to the networked communications solution, a tailor-made solution can be supplied to meet any requirements.

HiPath Xpressions V5.0 runs under a Windows 2003/XP Professional operating system in conjunction with these communication platforms:
- HiPath 2000
- HiPath 3000
- HiPath 4000
- HiPath 5000
- HiPath 8000
- HiPath DX
- Hicom 150 E/H
- Hicom 300 E/H

The following PBX-Connectivity’s are planned to be released soon:
- Cisco CM 5 (SIP)
- Alcatel OmniPCX Enterprise (SIP / ISDN)
- Nortel CSE 1000 (ISDN)
- Avaya S8300 (SIP)
- NEC 2400 IMX

Xpressions V5.0 can connect to either a HiPath 3000 v5 or a HiPath 4000 v2 via the CorNet-IP protocol. In addition, Xpressions V5.0 provides IP connectivity to the HiPath 8000 v2 via the SIP protocol. HiPath Xpressions employs the QSIG protocol to digitally connect to non-Siemens PBXs, providing MWI, calling party, called party, single B-channel transfers and forwarding condition (RNA/Busy). Connectivity to non-Siemens PBXs includes Avaya, Cisco Call Manager, Mitel, NEC and Nortel. HiPath Xpressions also supports SMDI connectivity to carrier-grade switches (e.g., Centrex).

Connectivity to non-Siemens communications systems may require HiPath Professional Services should testing to determine the extent of interoperability with a specific release be required.

7.1.0 Support for Open Standards

Vendor Response Requirement
Describe voice messaging system’s support for open standards.
List the clients that can be used with your proposed solution.
For proprietary clients, detail minimum hardware and software
Siemens Response: Comply
HiPath Xpressions architecture is a standards-based, comprehensive, unified messaging solution that adheres to standards such as MAPI, TAPI, IMAP4, HTML, CMC, X.400, TCP/IP, Ethernet, PPP, SMTP, LDAP and Java.

7.1.1 Security Features
Vendor Response Requirement
Describe security features available with the voice messaging system to prevent abuse and unauthorized access.

Siemens Response: Comply

HiPath Xpressions subscriber security varies depending on the access method — from the telephone user interface (TUI), browser-based tool (Web Assistant) or the graphical user interface (GUI) of the email client.

TUI security is provided by HiPath Xpressions through password controls. To log into the system, subscribers must provide both their telephone number and password using the telephone keypad. HiPath Xpressions ensures that the telephone and passwords match by using a one-way encryption algorithm.

Depending the account lockout policy, subscribers may be allowed multiple attempts to enter the correct telephone and password pair. The administrator also determines how long the account remains locked after the lockout has been triggered.

Subscribers can use the TUI or Web Assistant browser-based tool to set their passwords. The administrator determines password restrictions such as the minimum password length and password expiration period.

When subscribers access their messages from the GUI, the desktop application provides the security features.

For unified messaging configurations Exchange server topology and security will now apply to voicemail messages, therefore, Microsoft recommends a front end / back end server topology for organizations with multiple Exchange servers and users who access email via Outlook Web Access.

The front-end server can require secure socket layers (SSL) to access this front-end server.

7.2.0 Voice Mail Features

7.2.1 Forwarding
The system must provide access for forwarded calls from:
* Customer telephone system
* Direct central office (Business or Centrex lines)
* 800 Service lines

**Vendor Response Requirement**
Confirm support for each forwarding requirement.

Siemens Response: Comply

**7.2.2. Disconnect Detection**
The system should detect that a caller has hung up and immediately disconnect and restore the line to service.

**Vendor Response Requirement**
Confirm support for this operation.

Siemens Response: Comply

**7.2.3. Station Dialing**
In addition to the menu/route, callers may access an individual station either through the input of the extension number or the input of the called party’s last name. A total of 2,000 names plus 100 extension numbers will be possible.

**Vendor Response Requirement**
Confirm support for this operation.

Siemens Response: Comply

**7.2.4 Answer Announcement**
Individual, personalized announcements of 15-30 seconds for each mailbox user will be possible. A user's dictated answer message will only occupy the number of seconds dictated, with the remainder to be pooled so as to be available to:

1) all other mailbox owners; and,
2) for message taking.

Siemens Response: Comply

A system announcement of up to 30 seconds will be possible and also will be available in the event of switching system failure. It will be possible for the mailbox owner to input separate greetings for calls received internally or externally on the system. It will be possible for several individuals to share the same mailbox extension number. A caller reaching such a mailbox will be able to select between individual mailboxes.

**Vendor Response Requirement**
Confirm support for these operations.
Siemens Response:
1) “switching system failure” If the messaging system is installed behind the switching system and that switching system fails, a caller would not be connected to it or any systems behind it.
2) “internal or external callers” Siemens Response: Comply
3) Siemens Response: Comply, group or departmental mailboxes are supported.

7.2.5 DTMF Signaling
The system will be capable of receiving and generating standard DTMF tone signaling.

**Vendor Response Requirement**
Confirm support for this feature.

Siemens Response: Comply

7.2.6 Greeting
Voice mail calls will be answered on the first ring and be time- and date-stamped.

**Vendor Response Requirement**
Confirm support for this feature.

Siemens Response: Comply

7.2.7 Escape
A caller reaching the voice mail system will have the ability to re-route to an extension by dialing up to five digits or the operator by dialing "0" before or after leaving a message. It will not be possible for a caller reconnected to the telephone system to be connected to the public network.

**Vendor Response Requirement**
Confirm support for this feature.

Siemens Response: Comply

7.2.8 Trunk Access
It will be impossible for a caller passing through the attendant to reach an outside line.

**Vendor Response Requirement**
Confirm support for this feature.

Siemens Response: Comply

7.2.9 Distribution Lists
The system will contain a minimum of 80 distribution lists of at least 25 names each plus "all broadcast."

Vendor Response Requirement
Confirm support for this feature.

Siemens Response: Comply

7.2.10 Message Forwarding
Messages may be forwarded to single or multiple destinations with or without introductory comments.

Vendor Response Requirement
Confirm support for this feature.

Siemens Response: Comply

7.2.11 Audit Trail
It will be possible for a user to designate a necessary written record of message destination, input time and receipt. This audit trail will be printed on the administrative console together with daily reports.

Vendor Response Requirement
Confirm support for this feature.

Siemens Response: In addition to standard system reports, Siemens also offers professional services to fulfill any custom reporting requirements. The transactional database is a rich source of auditable data. It can be exposed to administrators for custom report writing using commercially-off-the-shelf reporting tools such as Crystal Reports.

7.2.12 Message Indication
The receipt of a message in a mailbox will cause a message-waiting lamp or "stutter" dial tone upon lifting of the station handset to indicate a message-waiting condition.

Vendor Response Requirement
Confirm support for this feature.

Siemens Response: Comply

7.2.13 Identification Code
Users accessing the system will input a discrete six-digit identification code which will be positively validated prior to access to their mailbox. Identification codes may be changed by mailbox owner.

Vendor Response Requirement
Confirm support for this feature.

Siemens Response: Comply

**7.2.14 Message Recovery**
The mailbox owner accessing the mailbox will be automatically told how many new messages have been received since last access and how many saved messages exist. Upon accessing the messages, the subscriber will have the choice of deleting, skipping or saving a message. Saved messages may only be deleted by the subscriber or by the system administrator.

**Vendor Response Requirement**
Confirm support for this feature.

Siemens Response: Comply.
1) The system will provide a tally of each message type and “read/unread” status when the mailbox owner accesses their mailbox.

**7.2.15 Message Reply**
A mailbox owner may respond to a message input by another system mailbox owner by simply depressing a single key.

**Vendor Response Requirement**
Confirm support for this feature.

Siemens Response: Comply. Mailbox owners can also be transfer to non-system mailbox owners assuming ANI is delivered with the original message.

**7.2.16 Message Review**
It will be possible for a user to review and edit either an announcement or input a message.

**Vendor Response Requirement**
Confirm support for this feature.

Siemens Response: Comply.

**7.2.17 User Controls**
A user accessing their mailbox will be capable of the following control functions:

1. Playback messages
2. Skip to next message
3. Cancel review
4. Replay last message
5. Replay faster or slower
6. Pause
7. Append information
8. Forward message (to mailbox or list)
9. Create new answer announcement
10. Increase play-back volume

**Vendor Response Requirement**
Confirm support for this feature. Indicate if any function is not supported.

Siemens Response: Comply.

**7.2.18 System Management Console**

The system will be equipped with a CRT and printer to provide system management functions. The administrative programs and traffic information secured will be possible during system operation. Traffic reports will be available on customer demand or automatically on a pre-programmed basis in quarter, half or one hour time frames or daily and weekly. At a minimum, they will indicate the following:

1. Storage space used for announcements or information mailboxes.
2. Storage space used for messages.
3. Maximum storage space used during the interval.

**Vendor Response Requirement**
Confirm support for this feature. Indicate if any requirement is not supported.

Siemens Response: Comply with clarification. Assuming the messaging system is accessible from the customer’s network, system administration functions as described are available from any network connected pc with the appropriate logon credentials.

**7.2.19 Traffic Reports**

Traffic reports will be available on customer demand or automatically on a pre-programmed basis in quarter, half or one hour time frames or daily and weekly. At a minimum, they will indicate the following:

1. Storage space used for announcements
2. Total calls answered
3. Total calls routed to station
4. Total calls routed to default
5. Total calls abandoned
6. CCS use and call count by input

**Vendor Response Requirement**
Confirm support for this feature. Indicate if any requirement is not supported.

Siemens Response: Comply. In addition to standard system
reports, Siemens also offers professional services to fulfill any custom reporting requirements.

**7.2.20 System Changeability**  
It will be possible for the system administrator to add and/or delete mailboxes, change general recordings and perform other administrative duties while the system is in operation.

**Vendor Response Requirement**  
Confirm support for this feature

Siemens Response: Comply

**7.3.0 Networking**  
VoiceCon plans on networking it new HQ messaging system to other VoiceCon locations equipped with messaging systems.

**7.3.1 AMIS**  
The proposed messaging system should support AMIS networking standards.

**Vendor Response Requirement**  
Confirm support for these features

Siemens Response: Comply

**7.3.2 Digital IP Networking**  
The proposed messaging system should support VPIM networking standards.

**Vendor Response Requirement**  
Briefly describe digital networking capabilities of your proposed messaging system solution. Indicate if VPIM is supported.

Siemens Response: Comply.

HiPath Xpressions communicates to enterprise messaging systems through connectors or gateways. The HiPath Xpressions digital networking does not depend on Exchange or any other groupware. HiPath Xpressions has its own networking protocol for digital networking between HiPath Xpressions sites regardless of the whether it operates in Exchange, Lotus Domino or GroupWise environment.

HiPath Xpressions uses the public and private networking facilities of the HiPath system, including LCR (least cost routing), for routing calls to and from the PBXs integrated with the voice mail systems.

Subscribers can send, receive and reply to messages from another AMIS and VPIM2 compatible voice mail system. HiPath
Xpressions Networking supports integrated networks of multiple HiPath Xpressions systems. With Connect Server(s) HiPath Xpressions will network with PhoneMail systems and support a PhoneMail LDN with network profile database and directory synchronization, dial by name and spoken name headers.

To insure disparate messaging systems are compatible, HiPath Professional Services may be required to determine that protocols supported by other vendors' messaging systems match those of Siemens' messaging Products, and that numbering plan issues are properly addressed.

7.4 Integrated Messaging Application

Vendor Response Requirement
Briefly describe how the proposed voice messaging system is to be integrated with VoiceCon's text messaging system, based on a MS Exchange server, to provide unified messaging system functionality. Station users must be able to view and access all messages (voice, text, fax) from their PC display monitor. Email text messages must be accessible from a telephone using text-to-speech conversion.

Siemens Response: Comply.
Efficient integration is available for Microsoft Exchange Server 2003.

The figure below shows a typical network diagram of a HiPath Xpressions unified messaging solution in an Exchange environment:
The Exchange Connector in HiPath Xpressions behaves like an Outlook client in a Microsoft Exchange server environment (i.e., provides store and forward capabilities). Outlook is accessed via the Exchange Connector APL, the interface between Outlook client and the Exchange server. A fully functional Outlook client must therefore be installed on the PC on which the Exchange APL should run. The Exchange Connector is installed on the HiPath Xpressions server. An additional connector is required for True Unified Messaging (single message store architecture), allowing customers to choose either dual message store or single message store.
8.0.0 Contact Center (Informational, only)

VoiceCon has future plans to install and operate a mid-size contact center solution across its three HQ locations. The contact center would integrate incoming voice, email, and web contacts from customers, and also support outgoing voice calls to potential customers. It is anticipated that the contact center will require 50 multifunction agents positions, and 5 supervisor positions. The contact center features and functions are NOT to be included in the configuration or pricing proposal.

8.1.0 Incoming Voice Call Center

The voice contact center solution should support call prompting, detailed call screening, and intelligent call routing capabilities. Agent groups should be both fixed and virtual based on skill profiles of the agents. Client/server CTI applications must be supported at all agent desks. Agent group assignments must be able to be distributed across the three HQ locations. The system should be designed to minimize agent requirements and call waiting times. Realtime supervisor reports and detailed historical reporting is required.

Vendor Response Requirement

Briefly describe you’re the system architecture of your incoming voice call center solution to satisfy VoiceCon’s basic requirements (see below). Include specific information about the system design architecture of your solution (hardware and software requirements), and specific capacity parameters for agents, supervisors, groups, announcements, queue slots, trunks and trunk groups, et al.

8.1.1 Basic Call Control Capabilities

At a minimum the proposed solution must be able to provide call control based on:

- ANI/DNIS
- call volumes
- performance criteria
- priority queuing

Vendor Response Requirement

Briefly describe the call control methodology used by your system that analyzes, routes, and queues calls based on each of the criteria.

Siemens Response: The HiPath ProCenter system utilizes a server-based architecture. Multiple software servers run the various processes in the system as illustrated in the figure below.
Server functions follow.

<table>
<thead>
<tr>
<th>Server</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>Administration Server</td>
<td>Controls access to the configuration information in the HiPath ProCenter Database through individual objects or summary collections</td>
</tr>
<tr>
<td></td>
<td>Provides object level access to the HiPath ProCenter Database even though the object may be physically stored in separate tables</td>
</tr>
<tr>
<td></td>
<td>Sends events when needed indicating configuration information has been changed</td>
</tr>
<tr>
<td>Call Director Server</td>
<td>Integrates with an Interalia XMU+ or Interalia SBX to interact with callers. Call Director collects customer-entered digits; provides menu support and plays messages and announcements</td>
</tr>
<tr>
<td>Callback Server</td>
<td>Controls access to callbacks stored in the HiPath ProCenter Database including the creation, modification and deletion of callbacks</td>
</tr>
</tbody>
</table>
|                        | Provides the ability to import a list of callbacks through the
<table>
<thead>
<tr>
<th>Server</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>Outbound Feature</td>
<td>Controls the scheduling and activity of all callbacks in HiPath ProCenter</td>
</tr>
<tr>
<td>Config Sync Server</td>
<td>Propagates switch configuration information to HiPath ProCenter for common resources (such as users, RCGs, ACD groups, MLHG's, user extensions, Call Director extensions and IVR extensions) based on the defined set of domain ranges. When synchronization is triggered, this server determines the configuration state of HiPath ProCenter resources by comparing the existence and configuration of these resources with the switch.</td>
</tr>
<tr>
<td>email Server</td>
<td>Communicates with the corporate email server using IMAP4 to process inbound messages and SMTP to send replies and outbound messages</td>
</tr>
<tr>
<td></td>
<td>Controls the movement and activity of all email messages in HiPath ProCenter</td>
</tr>
<tr>
<td></td>
<td>Maintains a contact model for the life of an email message in the system and sends events when the state changes</td>
</tr>
<tr>
<td></td>
<td>Provides an integrated auto-acknowledgement and auto-response mechanism for email messages, in conjunction with the workflow mechanism in the Routing Server</td>
</tr>
<tr>
<td>Real-Time Server</td>
<td>Provides formatted performance statistics to the Client Desktop application</td>
</tr>
<tr>
<td></td>
<td>Provides real-time and cumulative reports to the Manager application</td>
</tr>
<tr>
<td></td>
<td>Integrates with wallboards to display messages and statistics</td>
</tr>
<tr>
<td></td>
<td>Note that there may be more than one Real-Time Server in a HiPath ProCenter system, depending on the number of Client Desktops configured in a contact center. Auxiliary Real-Time Server software is installed on each HiPath ProCenter Auxiliary Server machines, as necessary to support the total number of users.</td>
</tr>
<tr>
<td>Reporting Server</td>
<td>Performs scheduling and generation of historical reports and provides data to the Manager application</td>
</tr>
<tr>
<td>Routing Server</td>
<td>Provides a workflow engine to process incoming voice, email and Web collaboration contacts through routing strategy and queue processing flows, and to process voice contacts through networking flows</td>
</tr>
<tr>
<td></td>
<td>Manages the routing and queuing of contacts for all media types (voice, email, callback and Web collaboration)</td>
</tr>
</tbody>
</table>
|                 | Manages the networking of voice contacts with other HiPath

©2005 TEQConsult Group
<table>
<thead>
<tr>
<th>Server</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>Server</td>
<td>ProCenter sites</td>
</tr>
<tr>
<td></td>
<td>Performs assignment of contacts to users based on either group-based routing or skills-based routing mechanism</td>
</tr>
<tr>
<td>SAP CIC Server</td>
<td>Provides integration between HiPath ProCenter and SAP CIC (as part of mySAP CRM) for voice and email media types</td>
</tr>
<tr>
<td>SAP ICI Server</td>
<td>Provides integration between HiPath ProCenter and SAP ICI (as part of mySAP CRM) for voice media</td>
</tr>
<tr>
<td>Statistics Server</td>
<td>Tracks user and contact-related data and saves the information in the HiPath ProCenter Database. Every 15 minutes, the Statistics Server rolls-up summary data. The Statistics Server also performs database maintenance, such as the purging of old data, and daily, weekly and monthly roll-ups of reporting data.</td>
</tr>
<tr>
<td>T-Server</td>
<td>Issues CSTA Service requests against the switch, and receives CSTA events from the switch, in order to support CTI functionality within HiPath ProCenter, through a CSTA provider (for all switches except the HiPath 8000), or directly through the switch for the HiPath 8000.</td>
</tr>
<tr>
<td></td>
<td>Maintains state information for all users, in all media (voice, email, callback and Web collaboration).</td>
</tr>
<tr>
<td>Watchdog Server</td>
<td>Starts, stops and monitors the status of all other HiPath ProCenter servers, and monitors the status of the Informix Database Server (main server machine only)</td>
</tr>
<tr>
<td></td>
<td>Sends status information (for example, a change in the status of a HiPath ProCenter server) to the IT Monitor application</td>
</tr>
<tr>
<td></td>
<td>Sends notifications (via pager or email) to inform the appropriate party of the status of, or potential problems in, the HiPath ProCenter system</td>
</tr>
<tr>
<td></td>
<td>Note that there may be more than one Watchdog Server in a HiPath ProCenter system. Each HiPath ProCenter Auxiliary Server machine used to provide additional Real-Time Server capacity will also have a Watchdog Server.</td>
</tr>
<tr>
<td>Web Interaction Server</td>
<td>Supports Web callback functionality: Integrates with a corporate Web server to collect web callback-creation data, and then supplies this information to the Callback Server in order to create a callback</td>
</tr>
<tr>
<td></td>
<td>Supports Web collaboration functionality: Integrates with a corporate Web server to enable the handling of Web collaboration contacts in HiPath ProCenter</td>
</tr>
<tr>
<td></td>
<td>Maintains a contact model for the life of a Web collaboration contact in the system and sends events when the state changes</td>
</tr>
<tr>
<td></td>
<td>Provides the capability to push messages and URLs, in</td>
</tr>
</tbody>
</table>
Auxiliary server machines will include Watchdog server and real-time server software. If SAP CIC Server / SAP ICI Server software runs on an auxiliary server machine rather than the Main Server, the auxiliary server machine will not have a real-time server.

A central reporting server may include a Watchdog server, an Administration server, a statistics server and a reporting server.

### 8.1.2 Advanced Call Control Capabilities
As an option the proposed solution must be able to provide call control based on:
- agent skills
- customer preference
- inbound and outbound call levels
- multi-media

**Vendor Response Requirement**
Briefly describe your system’s call control methodology that analyzes, routes, and queues calls based on each of the criteria.

Siemens Response: HiPath ProCenter Routing Strategy flows can look at the Service Level, number of agents logged on and in what state, calls in queue and wait times. The Performance Routing feature adjusts routing steps to reduce the amount of time contacts spend in queue.

A Service Level represents a performance statistic that indicates the percentage of contacts that are successfully answered within the service level interval. A Service Level Interval represents the amount of time (in minutes and seconds) within which a contact should be assigned to and accepted by an agent.

Administrators can choose one of four methods that use answered calls, and optionally abandoned and redirected calls, to calculate a Service Level. The meaning of the terms answered, abandoned and redirected, differ by media.

Administrators can specify one global Service Level interval for each media. This value serves as the default for a site. Within each queue definition, administrators can override the global service level with a custom interval to be used in calculating the service level for that queue.

### 8.1.3 Caller Notification of Wait Time
The proposed solution must be able to notify callers of expected wait times and “place” in queue and support information collection (such as an automated attendant feature) using “internal” hardware and software.

Siemens Response: Siemens Complies

**Vendor Response Requirement**
Describe how the application calculates wait time and any optional hardware or software required. Include a statement addressing if the announcement of wait time has an impact on a caller’s state in queue?

Siemens Response: The estimated wait time is calculated by dividing the number of calls in queue by the departure rate, where:
- The departure rate is the number of calls dequeued from the Routing Server that were associated with the specified queue. The departure rate is recalculated each time a call is answered, abandoned or transferred.
- The number of calls in queue equals the number of calls associated with the specified queue that are currently queued with the Routing Server. When a call of the same queue is enqueued to the Routing Server (regardless of the method used), this number increases by one.

Call Director includes performance messages to callers in queue. Wait time performance messages include current answer average and current answer estimated. Contact performance messages include contacts in queue and contact position in queue.

**8.1.4 Transfer to Voice Messaging Application**
After a configurable time, the caller should be able to transfer to a voice messaging system to leave a callback message.

**Vendor Response Requirement**
If the caller chooses to continue waiting rather than hanging up after leaving a message, describe how the call is placed back in queue.

Siemens Response: A caller can returns to HiPath ProCenter using menu prompts workflow provided by the a voice messaging mailbox. If desired, the workflow configuration can assign a priority level that returns the call to approximately the same place in queue as if the caller had remained in queue.
8.1.5 GUI Administration Tool
Supervisors must be able to reconfigure call control and assignments in real time, change priority of multiple calls simultaneously, view details of orphaned calls and retain customized settings regardless of log-on location. The solution must use a GUI administration tool and provide a graphical editor and what-if modeling as standard.

Vendor Response Requirement
Describe the system’s GUI administration tools.

Siemens Response: Supervisors can make minor changes to the production database (i.e., routine maintenance) in real-time. In addition to adding and deleting agent definitions, routine agent maintenance includes items such as assigning agents to new groups, changing the permissions and responsibilities of agent and so on. Other changes in this category include items such as changing a Call Director menu setup, adding service for a new dialed number and revisiting a queue/group setup.

Major reconfiguration changes such as changing flows should be made in the design database in order to validate changes prior to placing them in operation.

8.1.6 Soft Client
A soft client agent telephone and supervisor console will be highly desirable for both premises and off-premises locations.

Vendor Response Requirement
Describe the soft clients available for agent and supervisor use. The soft client must provide on-line help, ability to reserve calls or change call priority. For proprietary clients, detail minimum hardware and software requirements.

Siemens Response: The Client Desktop provides a standard, intuitive interface for handling all types of customer contacts, whether they are voice calls, voice callbacks, email messages or Web collaboration sessions. Certain options and permissions are set in the Manager application, and these settings determine which features are available to an individual agent in the Client Desktop.

The main window (see figure below) remains on an agent’s screen, in either full or icon mode, for the entire time that the agent is working in the Client Desktop.
Client Desktop includes features that contain commands (e.g., Menu bar, Standard toolbar and Media bar) and features that receive information (e.g., Broadcaster, Personal Performance toolbar and Status bar and system messages window). Some toolbars (e.g., Team Bar and Team List) provide both information and access to commands using a submenu.

Team List allows agents to identify the status and availability of peers, supervisors and ‘extended’ contact center agents. Listed users can be configured by the Administrator or Supervisor, with up to 25 users per list. Agents can call, transfer, consult or conference with team members via popup menu or context menu.

The Team Bar provides:
- A shortcut ‘buddy list’ from Team List, configurable by agent
- A convenient visual monitoring of presence and availability interface, including work and unavailable reasons
- Single click access to invoke a call

The main window contains the following items:
- **Title bar** — displays the name of the system (HiPath ProCenter) and the component running
- **Menu bar** — contains the File, Edit, View, Tools, Actions, Windows and Help menus familiar to Windows GUI users
- **Standard toolbar** — contains buttons clicked to execute commands. The buttons provide an alternate method of working, depending on a user’s preference. Frequently used commands are accessible from both the menu bar and the standard toolbar.
- **Broadcaster** — displays information and messages about the contact center
- **Personal Performance toolbar** — displays an agent’s Personal Performance statistics (e.g., Utilization, Average Voice Handle Time and Average Work Time)
- **Speed Bar** — provides quick access to frequently called numbers in the Speed List
- **Team Bar** — provides quick access to frequently called numbers in the Team List
Media Bar — provides easy access to frequently used functions, such as displaying the Personal Performance dialog box, creating a new callback or e-mail message and performing an email history search

Status bar — displays system messages, and on the right side, user name.

There is usually more than one way to perform an action in the Client Desktop. For example, an agent can click an option on a menu, click a button on a toolbar or right-click an item and then click an option on the small menu that appears.

If a button is dimmed, it is not available to the present time. Regardless of how a button appears, a ToolTip showing a button’s name or function appears when the user points the mouse at it.

8.1.7 ACD Voice Terminal
IP desktop voice terminal instruments will be required for agent positions.
Vendor Response Requirement
Briefly describe any telephone instruments designed specifically for ACD agents. Include any and all feature/function attributes unique to ACD operations. Provide a photograph of the instrument, if available.

Siemens Response: The Siemens OpenStage telephones can be used for desk top agents. See previous section on phones. The phone is programed with the appropriate features for a HiPath ProCenter Agent.

8.1.8 Supervisor Real-time Call Handling and Performance Status
Supervisor terminals must show, in real time, all logged-on agents, the status of each agent, caller queue information and thresholds and alarms. Users must be able to customize displays.

Vendor Response Requirement
Describe the proposed solution's real time supervisor console display capabilities for assisting supervisors with managing the customer interaction center. Include a diagram illustrating two or three screen displays available to the supervisor.

Siemens Response: Real-Time statistics are available on queues, individual agent (users), virtual groups and or a collection of resources, called departments or aggregates. Supervisors can customize real-time views to include only those statistics desired, and can decide whether to include tabular or graphical information, or both. Supervisors can also combine up to four views on the same screen (see figure below).
System administrators configure system thresholds. Each agent and supervisor can configure threshold behavior (e.g., color, animation, audible alarm) on the Client Desktop or HiPath ProCenter Manager application, respectively. In addition, report definitions can include thresholds (see figure, right). Column values are continually updated and visible or audible alarms can draw attention to certain conditions.

Each real-time report subtype provides a set of columns for which alarms can be set. When setting up a report, supervisors can define up to three different thresholds on a particular column. A different alarm can be displayed
or sounded whenever each threshold value is exceeded.

For example, a key piece of information in a contact center is the number of contacts currently waiting in queue. To monitor contacts waiting at the queue level, supervisors use a queue real-time report. The real-time Queue Status Report shows current contact waiting totals for a set of queues.

Details for each queue include:
- The number of contacts waiting in that queue
- The current service level (based on the last 24 contacts) and an estimated service level
- The estimated time that a contact will wait before being answered

<table>
<thead>
<tr>
<th>Name</th>
<th>Contacts In Queue</th>
<th>Service Level</th>
<th>Wait Time</th>
<th>Service Level</th>
<th>Estimated</th>
<th>Current Answer Estimated</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mutual Funds-DIV</td>
<td>0</td>
<td>100.0</td>
<td>100.0</td>
<td>00:00:23</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Mutual Funds-EQU</td>
<td>1</td>
<td>83.0</td>
<td>77.0</td>
<td>00:02:53</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Stocks</td>
<td>2</td>
<td>100.0</td>
<td>83.0</td>
<td>00:02:45</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>81.0</td>
<td>86.0</td>
<td>00:01:00</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Figure 4 — Queue Status Report (Group-Based Routing)**

The highlighting on the Contacts: In Queue column is an alarm mechanism that provides a visible or audible alarm. Each real-time report subtype provides a set of columns for which users can set alarms. When setting up a report, the supervisor can define up to three different thresholds on a particular column. A different alarm can be displayed or sounded whenever any threshold value is exceeded.

In a real life situation, the exception might mean nothing more than an unusually busy day. A supervisor might use this information to take immediate action — have more users become available, for example. Alternatively, a supervisor might use a queue status report as the first step in tracking down a problem, combined with a user real-time report on users associated with that queue for additional details.

**Figure 5 — User Real-Time Report (Group-Based Routing)**
8.1.9 Agent Display Information

Vendor Response Requirement: Describe real-time display information provided to agents at their desktop via their hard telephone instrument and the softclient solution.

Siemens Response: HiPath ProCenter agent displays focus on primarily on real-time statistics. Alarms are highlighted in the user-configured color (generally red) to emphasize areas for improvement.

The Client Desktop provides information about performance information in multiple ways.

- Broadcaster content can continue the time and date, simple text messages and a selection of HiPath ProCenter performance statistics.
- Personal Performance information provides agent-specific information (see figure below).

Agents can use the Personal Performance dialog box to customize the Personal Performance toolbar — for example, to add statistics to the toolbar and to order the statistics on the toolbar. Agents can also choose to view Personal Performance in a vertical list rather than a scroll bar to view more statistics.
The following paragraphs detail information available at the desktop or on wallboards. Client Desktop statistics can include user (agent), contact, group and queue information.

Agent performance statistics (available in Client Desktop views) include:

<table>
<thead>
<tr>
<th>State — Time</th>
<th>Utilization</th>
</tr>
</thead>
<tbody>
<tr>
<td>Average Time — Routed Handled</td>
<td>Total Time — Handled</td>
</tr>
<tr>
<td>Average Time — Direct Handled</td>
<td>Percentage Time — Available</td>
</tr>
<tr>
<td>Average Time — Held</td>
<td>Percentage Time — Unavailable</td>
</tr>
<tr>
<td>Average Time — Work</td>
<td>Percentage Time — Work</td>
</tr>
<tr>
<td>Handled — Routed</td>
<td>Percentage Time — Routed Handled</td>
</tr>
<tr>
<td>Handled — Direct</td>
<td>Percentage Time — Direct Handled</td>
</tr>
</tbody>
</table>

Contact performance statistics (available in Client Desktop views) include:

<table>
<thead>
<tr>
<th>Source</th>
<th>Queue</th>
</tr>
</thead>
<tbody>
<tr>
<td>Destination</td>
<td>Wait time</td>
</tr>
<tr>
<td>Calling Extension</td>
<td>Description</td>
</tr>
</tbody>
</table>
**Group and virtual group performance statistics** (available in Client Desktop and wallboard views) include:

<table>
<thead>
<tr>
<th>Calls Waiting — All</th>
<th>Agents — Unavailable</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calls Waiting — Primary</td>
<td>Agents — Working</td>
</tr>
<tr>
<td>Calls Waiting — Overflow</td>
<td>Agent — Handling Routed</td>
</tr>
<tr>
<td>Agents — Logged On</td>
<td>Agents — Handling Direct</td>
</tr>
<tr>
<td>Agents — Available</td>
<td></td>
</tr>
</tbody>
</table>

**Queue performance statistics** (available in Client Desktop and wallboard views) include:

<table>
<thead>
<tr>
<th>Abandon Rate — Current</th>
<th>Service Level — Current</th>
</tr>
</thead>
<tbody>
<tr>
<td>Abandon Rate — Shift</td>
<td>Service Level — Estimated</td>
</tr>
<tr>
<td>Answer Rate — Current</td>
<td>Service Level — Shift</td>
</tr>
<tr>
<td>Answer Rate — Shift</td>
<td>Wait Time — Oldest Contact</td>
</tr>
<tr>
<td>Contacts — Received</td>
<td>Wait Time — Current Abandon Avg.</td>
</tr>
<tr>
<td>Contacts — In Queue</td>
<td>Wait Time — Shift Abandon Avg.</td>
</tr>
<tr>
<td>Contacts — Answered</td>
<td>Wait Time — Current Answer Avg.</td>
</tr>
<tr>
<td>Contacts — Redirected</td>
<td>Wait Time — Shift Answer Avg.</td>
</tr>
<tr>
<td>Contacts Redirected — Out of Scope</td>
<td>Wait Time — Current Answer Est.</td>
</tr>
<tr>
<td>Contacts — Overflowed</td>
<td>Contacts — Networked In</td>
</tr>
<tr>
<td>Contacts — Abandoned</td>
<td>Contacts Redirected — Networked Out</td>
</tr>
</tbody>
</table>

**8.2.0 Reporting**
VoiceCon requires call center system operation reports in various formats.

**8.2.1 Statistical and Configuration Reporting**
VoiceCon requires sophisticated reporting to track and further enhance its CIC operations. Reports must be available on terminal display and paper printout and be able to be downloaded to a PC. The proposed solution must provide open storage capability.

**Vendor Response Requirement**
Describe the number of and type of information standard statistical, configuration and audit reports provided.

*Siemens Response: Report Center*

HiPath ProCenter’s Report Center is based on a powerful reporting engine to define and view a virtually unlimited number of real-time, cumulative and historical reports for all media. The flexible interface makes customizing reports easy without requiring an external report writer. Report Center provides insight into contact center operations, allowing for better operational monitoring, more effective decision making and the ability to proactively spot patterns and respond — before they become problems.

**Real-time and Cumulative Reporting**

Real-time statistics are available on users, groups, queues and callbacks. Supervisors can customize real-time views to include only those statistics desired, and can decide whether to include tabular or graphical information, or both. Supervisors can also combine up to four views on the same screen.

A built-in analytic model uses actual data trends to predict performance patterns and contact volumes in real-time, improving decision making regarding staffing resources or contact routing. Thresholds and alerts can easily be defined to provide audio and visual notification to a manager when definable operating metrics are exceeded.

Real-time and cumulative views are refreshed continuously, presenting key information such as agent utilization, service levels, abandon rates and average handling time for voice, email and callback interactions.

The HiPath 4000 supports the whisper announcement feature with HiPath ProCenter to provide silent coaching capability. Whisper announcement allows a supervisor to speak to an agent while blocking the caller from hearing what is said. The supervisor selects the agent by entering either the extension number or Agent ID, depending on the system configuration. Both the supervisor and the agent must include monitoring classmarks.

**Activity Logs**

Detailed, searchable activity logs allow managers to examine the step-by-step progression of any customer contact or review the detailed activities of an agent throughout the day for all media.

**Historical Reporting**

Historical Reports can be quickly created by just pointing and clicking to select data elements and report parameters. Managers can choose from a comprehensive range of statistical values for blended as well as media-specific reports.

Report Center provides graphical as well as tabular historical reports. Displaying historical reports in HiPath ProCenter’s Report Viewer allows managers to flexibly adjust report output even after the reports have been run. They can reorder and resort content, as well as tailor the level of detail displayed on screen.

Multiple reports can be opened in the Report Viewer concurrently, and are accessible with just a mouse click. Reports can be viewed on-demand or scheduled to run on a
daily, weekly or monthly basis. Output options include printing or export to Excel, HTML, PDF or text.

8.2.1 Graphical Reporting
The proposed solution must provide graphical reports as a standard feature.

Vendor Response Requirement
Describe the available graphical reports with your system.

Siemens Response: HiPath ProCenter’s Report Center is based on a powerful reporting engine to define and view a virtually unlimited number of real-time, cumulative and historical reports for all media. The flexible interface makes customizing reports easy without requiring an external report writer.

Customizable real-time, cumulative and historical reports are available in graphical and tabular format.

8.2.2 Call-by-Call Reporting
The proposed solution must provide call-by-call reporting as an optional feature.
Vendor Response Requirement
Describe your system’s call-by-call reporting capabilities, if available.

Siemens Response:
Activity reports, provided as a standard HiPath ProCenter feature, are useful to provide historical, detailed state-change data on a given agent for a particular interval or on a contact or contacts from a given source for a given day.

Supervisors can use User Activity Reports to investigate minute-by-minute agent actions. Specifically, supervisors can use an activity report to view:
- A step-by-step history of agent state and handling state changes for an agent over a given interval (see figure below)
- A step-by-step history of handling state changes for all email or voice contacts from a particular source over a given interval. For example, if a customer complains about excessive hold time or queue time on a particular contact, an activity report can provide specific details on that contact.
Figure 7. — User Activity Report

<table>
<thead>
<tr>
<th>Time</th>
<th>Activity</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>2/14/2005</td>
<td></td>
<td></td>
</tr>
<tr>
<td>16:00 To 17:00</td>
<td></td>
<td>15 Entries</td>
</tr>
<tr>
<td>4:29:57 PM</td>
<td>Logon</td>
<td>Logged on to: E-mail</td>
</tr>
<tr>
<td>4:29:59 PM</td>
<td>Offered contact</td>
<td>Contact Type: Routed E-mail, Que...</td>
</tr>
<tr>
<td>4:32:28 PM</td>
<td>User Replied Contact</td>
<td></td>
</tr>
<tr>
<td>4:35:01 PM</td>
<td>Logon</td>
<td>Logged on to: Callback</td>
</tr>
<tr>
<td>4:35:01 PM</td>
<td>Logon</td>
<td>Logged on to: Voice</td>
</tr>
<tr>
<td>4:37:07 PM</td>
<td>Submitted Wrap-up</td>
<td>Reason: Schedule MRI</td>
</tr>
<tr>
<td>4:37:27 PM</td>
<td>Offered contact</td>
<td>Contact Type: Routed Callback, Que...</td>
</tr>
<tr>
<td>4:37:27 PM</td>
<td>Answer contact</td>
<td></td>
</tr>
<tr>
<td>4:37:52 PM</td>
<td>Working</td>
<td>Reason: Mandatory</td>
</tr>
<tr>
<td>4:37:53 PM</td>
<td>Disconnect contact</td>
<td></td>
</tr>
<tr>
<td>4:38:05 PM</td>
<td>Logoff</td>
<td>Logged off of: E-mail</td>
</tr>
<tr>
<td>4:38:05 PM</td>
<td>Logoff</td>
<td>Logged off of: Callback</td>
</tr>
<tr>
<td>4:41:45 PM</td>
<td>Offered contact</td>
<td>Contact Type: Routed E-mail, Que...</td>
</tr>
<tr>
<td>4:52:21 PM</td>
<td>Submitted Wrap-up</td>
<td>Reason: Schedule MRI</td>
</tr>
<tr>
<td>4:53:11 PM</td>
<td>User Replied Contact</td>
<td></td>
</tr>
</tbody>
</table>

[Deleted: 4]
8.3.0 Self Service

The proposed solution must support self service (e.g., IVR) integration as an option. Callers must be able to retain their place in queue while using IVR features.

**Vendor Response Requirement**

Describe your system’s ability to support inbound calling, call control services, messaging for agents, speech recognition, text-to-speech, TDD and CTI and integration with a customer self-service interaction application.

Siemens Response: The optional Call Director feature provides front-end menus, announcements such as position in queue or estimated wait time and collects digits that have been input by callers for identification purposes. Call Director supports up to 64 ports and 6 languages for TDM or IP environments.

Call Director:

- Enables automated front-end call processing without the complexity of IVR programming through a simplified, graphical user interface
- Facilitates simple self-service and automated transactional call processing to improve customer access and convenience while lowering operating costs
- Improves customer satisfaction through timely greetings, intelligent announcements (e.g., expected wait time) and gathering specific call routing criteria using dynamic, multi-format numbers-to-speech playback
- Speeds up call resolution and streamlines customer service through automated collection of caller information (e.g., account number, PIN number)
- Enhances agent productivity by providing previously collected customer data synchronize with the incoming call for HiPath ProCenter Agent or third-party application (e.g., Microsoft CRM)

HiPath ProCenter Call Director even allows for simple interactive call flows, combining digit collection from the caller with read/write access to open ODBC databases with customer data and number playback functionality, facilitating the execution of simple transactional and self-service applications without involving a live agent. HiPath Professional Services are necessary for more complex Call Director implementations.

Call Director provides 20 performance messages for a queue. Messages related to estimated wait times include wait time for current answer average, current answer estimated, oldest contact, current abandon average, shift answer average and shift abandon average; current abandon rate; contact position in queue and contacts in queue.

In addition to significantly lower implementation costs, Call Director can be deployed concurrent with the HiPath ProCenter implementation. Customers who require database query for automated servicing, more complex menus, more capacity or forget-me-not queuing should deploy an IVR rather than Call Director.
8.3.1 Script Development
Vendor Response Requirement
Describe the design tools/environment for IVR script development, the method used to test applications and changes prior to putting them into production and the method of putting changes into production.

Siemens Response: Administrators use Design Center, fully integrated into the HiPath ProCenter Manager application, for programming contact center routing. Design Center programs queues (logical holding places for incoming contacts) and flows (programs that dictate how incoming contacts are processed and what callers in queue hear and the actions they can take).

The visual workflow creation tool:
- Provides an easy-to-use single point of design for all routing strategies, integrated self-service application and queue processing flows
- Includes a library of configurable, reusable routing and queue processing components
- Checks and validates routing strategy and queue processing flows for completeness in real-time
- Provides visual definition of multi-site networking workflows

Design Center provides a intuitive, workflow-style design tool for multimedia routing strategies (see figure below).

![HiPath ProCenter Design Center](image)

**Figure 8 — HiPath ProCenter Design Center**
For example, when designing Voice Routing, Design Center uses conditional and data directed routing strategies assign caller to the most appropriate queue, where contacts wait until handled by an agent. Contacts can overflow to subsequent group or virtual skills group based on elapsed time in queue. Design Center also easily integrates Call Director IVR components within a voice routing workflow.

Design Center flow components provide a wide range of functionality that lets contact center managers to collect information on incoming contacts, make routing decisions and otherwise automate aspects of a contact center. While most of the operations and functionality are contact center-specific, the programming mechanism provides common programming functionality, such as basic sequential processing, decision making, calls to other flows and looping. In addition, administrators can use the functions provided to automate aspects of contact center operations. The GUI minimizes configuration burdens through:

- Configurable and reusable components for building routing strategy and queue processing flows
- A point-and-click approach to the configuration of flows, allowing administrators to make changes ‘on-the-fly’
- Automatic validation of workflows
- The capacity to store flows ‘off-line’ for later or remote access (when not logged on to a production database)
- The ability to print or export call flows as documentation

Each flow is a visual script created using the Manager application’s Design Editor configuration tool. With this tool, administrators create flows using a method similar to creating a programming flowchart (see figure below).
The small square objects are flow components. Each component performs a specific action and the links, shown as arrows, dictate the order in which the actions occur. Where a component has more than one exit point leading away from it, that component performs a test and the next component to execute depends on the results of the test. The different icons on the components indicate different component types. Different component types perform different actions.

Administrators can add a component using the “Add Components” dialog box (similar to adding a shortcut to My Favorites). If a component cannot be used in the current flow, the application indicates this by displaying a “cannot place component” icon when the user drags the component onto the flow diagram.

The Component Shortcuts area contains frequently-used components that have been manually added. Administrators can drag a component from the component library to the Component Shortcuts area, making them readily available.

Design Center also uses pre-configured screens to design individual routing strategy and queue workflow components.
**Custom Routing**

Administrators can write a custom function in a DLL or COM module and call the function from a routing strategy or queue processing flow, for all supported media. The function is passed contact data associated with the contact and the function can:

- **Modify provided contact data values.** For example, if the workflow prompts the customer for a seven-digit account number, a custom function could format the number to include spaces or dashes, for display purposes.

- **Add new contact data.** This would be useful in making use of a value not normally available in a flow. This value might be derived from information passed as inputs, by a calculation for example. Similarly, the information might be obtained from a third party source (e.g., by database lookup), processed in some way and passed back to the flow. Once the custom flow finishes executing, the new information is available as contact data for use in the flow, to be used in routing decisions or for display in the Client Desktop.

- **Perform tasks outside of the scope of HiPath ProCenter processing.** Once a custom function has been written and compiled, HiPath ProCenter can call the function in a routing strategy or queue processing flow.


An example of this is writing a customer-entered seven digit account number to the contact data. The custom function parses out the third, fourth and fifth digits, corresponding to the caller’s membership level, and creates a new contact data key/value pair with these digits. The contact could then be enqueued based on whether the three digits constitute a premium or general membership.

**8.4.0 Workforce Management System**

The proposed solution must provide forecasting and scheduling capabilities as an option.

**Vendor Response Requirement**

Describe your system’s workforce management capabilities.

Siemens Response:  *Workforce management integration* uses XML to export information about users, groups, user activities, and voice contact statistics for use by third-party workforce management applications, such as Witness (previously Blue Pumpkin).
8.5.0 Integrated Email Call Control
The proposed solution must integrate customer email messages as an option. It is also desirable that agents be able to handle a mix of voice and email messages on a call-by-call basis, and that all incoming voice calls and emails be routed into the same agent queue(s).

Vendor Response Requirement
Describe your system’s capability to integrate email contact center functions with your voice call center system. Include information about the hardware and software requirements for this application.

Siemens Response: HiPath ProCenter supports the following email and LDAP components:
- **Protocols**: IMAP Version 4 and SMTP Version 1 (Enhanced SMTP is not supported)
- **Servers**: Microsoft Exchange 2000/2003 and IBM Lotus Domino Mail Server 6.5/7.0

8.6.0 Web Center
VoiceCon anticipates that it will require integration of its call center with its web server system. The proposed solution must support customer-initiated contact through the Internet as an option.
Vendor Response Requirement

Describe how your call center can be integrated with the VoiceCon website to allow agents to respond to customer callback requests via the website. Include in the discussion whether agents can collaborate in realtime with callers during an online website transaction.

Siemens Response: A full set of telephony controls and tools streamlines handling incoming calls, email contacts, agent and web initiated callback interactions and (HiPath ProCenter Enterprise only) web collaboration contacts. Synchronized with the arrival of each interaction at the desktop, the agent receives a “screen pop” with customer data and contact details appropriate to the contact media. In addition, an interface to 3rd party or in-house CRM systems can be used to automate customer file retrieval for display on the agent's screen.

Agent Desktop insures superior customer service and reduces the need for later agent follow-up. Benefits of the universal agent desktop application include the following:

- Presence and collaboration tools allow agents to leverage other resources to increase first contact resolution
- Blended contact handling allows agents to blend media during customer interactions, creating richer customer contacts and improving quality of service.
- Agents can initiate emails to customers while being on a call or in a web collaboration session to make additional information accessible beyond the real-time interaction.
- Agents can create callbacks based on incoming emails, calls or web collaboration interactions, ensuring reliable and timely follow-up.
- Agents can initiate calls while processing incoming emails or web collaboration interactions.

8.7.0 Outbound Dialing

The proposed solution must support automated outbound predictive dialing as an option.

Vendor Response Requirement

Describe your system’s capabilities to perform outbound predictive dialing, and include necessary hardware/software requirements.

Siemens Response: HiPath ProCenter, although targeted for contact centers with primarily inbound contacts, has a very rich Outbound, proactive callback option that is very cost effective for customers, doesn't require additional hardware and is fully integrated with the HiPath ProCenter Agent Desktop application. This solution offers preview-dialing capabilities and delivers higher levels of agent utilization while providing an accelerated ROI in contact centers that are primarily inbound centers but require the benefits of true call blending.
Callbacks are handled similar to preview dialing, where the agent sees the information about the caller first, and presses a button to call that customer. Once a callback is enqueued, it is handled very much in the same manner as an inbound telephony call. It is placed in the same queue as inbound calls. It is routed by the same sophisticated intelligent routing engine based on callback requirements and agents skills.

For customers that wish to employ third-party outbound dialing solutions that support predictive dialing, Siemens offers open functionality provided with our Software Developers Kit (SDK).

8.8.0 Server-Based CTI Call Control
The proposed solution must support server-based CTI applications as an option.

Vendor Response Requirement
Describe the capabilities of the proposed solution to simultaneously route a call and data screen populated with the caller's identity, location or reason for calling.

Siemens Response: The CTI gateway server on which the HiPath CAP application resides performs the protocol conversion and allows the HiPath 8000 to communicate with telephony applications residing in the computer system environment. The HiPath 8000 supports up to sixteen simultaneous CTI links.

HiPath ProCenter supports two CTI applications — Agent Desktop and Attendant Console.

Integrated screen pops provide:
- Key contact information, including actual caller wait time in queue
- Contact data, which may include collected digits from Call Director or an IVR and information gathered from a database lookup in the call flow
- LDAP Directory Lookup information
- Wrap-up reasons the agent selected during or after the call (Agent Desktop only)
- The ability to select which screen the agent sees first (Agent Desktop only)
- Line Status for Agent Desktops (Attendant Console only)
- Extended History (Attendant Console only)
- A configurable setting controlling the number of Contact Details windows open.
In addition to the integrated screen pop, HiPath ProCenter supports a screen pop API for customer-created application, SDKs for integration with SAP CIC/ICI Desktops and Microsoft CRM integration.

HiPath Professional Services may be necessary to customize standard Contact Details functionality.
Part 2: System Pricing

1.0 System Pricing Requirements

Summary system and voice terminal pricing data will be presented to VoiceCon workshop attendees and be deemed for public use.

Detailed pricing data will remain confidential, and used to verify if the proposed system configurations satisfy RFP requirements.

Installation fee pricing data is required, and must be included in the RFP response. Indicate if the proposed installation fee is based on direct sales/service or a channel partner pricing schedule.

The proposed system price must also include a 1-year warranty to the customer. If this is a pricing option in your pricing schedule include it as part of the installation fee, and identify it as such.

2.0 Summary Pricing – VOICECON NETWORK (all five locations)

Complete the attached EXCEL data table for your proposed system pricing summary data. The submitted data will be made available to the general public.

System Summary Pricing

All Common Equipment
(call processing, port interfaces, media gateways, housings, power, feature/application servers, et al)

Generic Software (Standard Features)

Optional Software (Including License Fees)

Desktop Voice Terminals

Systems Management/Administration System

Messaging System

Installation Fee (including 1-year warranty)

TOTAL
3.0 Desktop Voice Terminal Pricing

Complete the attached EXCEL data table for your proposed Desktop Voice Terminal pricing summary data. The submitted data will be made available to the general public.

**Voice Terminals**

- Economy Desktop IP Telephone Instrument
- Administrative Desktop IP Telephone Instrument
- Professional Desktop IP Telephone Instrument
- Executive Desktop IP Telephone Instrument
- IP Audio conferencing Unit
- PC Client Softphone (Station User) License Fee
- PC Client Softphone (Attendant) License Fee
- Key Module Add-on
- Gigabit Ethernet Module Add-on
- Display Module Add-on
- WLAN Module Add-on
- Desktop Power Module Option

4.0 Detailed Configuration Components and Pricing

Submit a separate EXCEL file with a detailed listing of proposed communications system components/elements and associated unit pricing, also indicating the proposed unit quantities included in the configuration for the base system (HQ facility). Also include an additional section with the configuration hardware/software elements and associated pricing data to satisfy each of the remote facilities (small, medium, large). Provide English language descriptions of all price configuration system components and elements in addition to any proprietary order codes.

At minimum, the configuration component list should contain:

- All common control elements
- All common equipment port cabinets/carriers
- All port circuit interface cards for station and trunk ports
- All media gateway equipment for station and trunk ports
- All call control signaling interface cards
- All voice terminals, including audioconferencing units
- Generic software
- All port license fees
- All optional software packages
  o Include all optional adjunct server equipment to support of required features
  o All voice messaging system elements (cabinet equipment and memory storage)
- All systems management elements

The detailed pricing file will NOT be made public, but will be used to verify adherence requirements and the pricing summary data.