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RECEIVED
OFFICE OF PUBLIC ACCOUNTABILITY
PROCUREMENT APPEALS

DEC 17 2010

TIME: 3:15 pm BY: JB
FILE NO. OPA-PA: 10-005

Attorney for Appellee, Guam Community College

BEFORE THE OFFICE OF PUBLIC ACCOUNTABILITY
PROCUREMENT APPEAL

In the Matter of Appeal of PACIFIC DATA SYSTEMS, INC, Appellant	Docket Number: OPA-PA-10-005 **SECOND SUPPLEMENTAL PROCUREMENT RECORD**

Appellee Guam Community College, (hereinafter "GCC"), by and through its attorney, Sarah A. Strock, of Cabot Mantanona LLP, respectfully submits supplemental procurement record relating to TeleGuam's bid:

1. Page 26 of section 7 (only the portion pertaining to tax)
2. Pages 30 and 31 (only the portions concerning tax) of section 10
3. Pages 84-108 of section 7
4. Pages 1-14 of the Cisco Unified Communications Manager Version 7.1 data sheet of section 12

   *The OPA's decision said this was found in section 11, but it was really in section 12.
5. Pages 16-19 of section 7
6. Page 23 of section 7
7. Pages 22-25 of section 10

8. Pages 28-29 of section 10

9. Page 26 if section 7 (as it pertains to commercial terms)

10. Pages 27-29 of section 7 (as it pertains to commercial terms)

    *Pages 30-31 of section 7 are forthcoming.

11. Pages 32-35 of section 10

Respectfully submitted this 17th day of December, 2010.

CABOT MANTANONA LLP
Attorney for Guam Community College

By: ______________________________
SARAH A. STROCK

TAX

The price includes GRT and any applicable federal mandated surcharges and fees.

VoIP DETAIL COST BREAKDOWN

EXHIBIT 2

9620 IP Telephone

A member of the Avaya one-X® Deskphone Edition family, the Avaya 9620 IP Telephone is specifically designed for the everyday telephone user—those who rely on multiple communications tools such as e-mail and IM, yet still require a high quality and intuitive telephone for voice communications.

Avaya 9620 IP Telephone

The 9620 IP Telephone features a 3.45 inch (9 cm) diagonal monochrome backlit display - which has been enhanced with higher resolution (1/4 VGA) compared with other available monochrome telephones from Avaya. The 9620 supports up to 12 call appearances and/or administered feature keys - with three concurrent line appearances visible at any time.

The 9620 features several LED lights and buttons. LED lights on the side of the display provide explicit status of different line appearances, while LEDs built into several buttons on the phone such as Mute, Message, and Headset provide an intuitive and simple experience for the everyday end user.

The user interface on the 9620 is helpful and intuitive, so completing call transfers and setting up ad hoc conference calls is simple and can be executed with confidence - even for the casual, everyday user.

EXHIBIT 3

Tax

The price includes GRT and any applicable federal mandated surcharges and fees.

Key Feature	Benefit
Intuitive User Interface – the intuitive, context-sensitive interface on the 9620 is designed to facilitate confident usage by featuring context-driven menus with on-screen prompts, facilitating straightforward access to the contact directory and call log. Everyday users have full control of conference calls, including selective drop and mute.	Easy access to common features such as Conference, Transfer, and Hold are greatly enhanced with the helpful prompts and tight integration with phone numbers in the contact list and call logs. The user interface on the 9620 provides easy access to critical Communication Manager features, and ultimately increases the end user's confidence with the phone and their productivity.
Superior Audio Quality – the unique high-fidelity acoustics of the Avaya 9620, including a full duplex speaker and wideband audio in the handset, deliver industry-leading audio that minimizes ambient noise. With the enhanced audio across high and low frequencies, it is easier for users to better understand others with different speech nuances or accents.	Because of the enhanced audio, calls are more productive, team members are better able to collaborate. On conference calls, users find it easier to distinguish and understand multiple speakers, aiding in collaboration and communications. Overall, communications are richer.
New design and display – the 9620 features an improved, higher resolution displays - supporting ¼ VGA gray scale with backlighting. A four way navigation button cluster is another new addition to the 9620 - providing a familiar, cell phone-like interface for navigation and feature selections for the everyday user.	The new design facilitates better usage of the display and the built in browser to improve access to information and use of telephone features.
Investment Protection - Modules and Adapters – based on open standards with a modular platform, the Avaya 9620 allows enterprises to add a wide range of modules and adapters to further enhance employee productivity. This facilitates individual choice between Bluetooth, monaural - wideband or binaural - wideband headsets. A standard USB interface accommodates a range of USB devices. In the future, additional adapters and modules including those for Gigabit Ethernet and Bluetooth will be available.	Investment protection and reduced total cost of ownership are key benefits of the 9620. Traditionally, the only way to receive new telephone capabilities was to purchase a completely new phone. Designed with the future in mind, the Avaya 9620 provides a flexible approach for adding future functionality to current telephones.

Key Feature	Benefit
9620 Designed just for the everyday user -- traditionally, desk phone design has focused on a paradigm of buttons and size of display. The higher the level of the intended user meant more buttons and larger displays - regardless of the functional needs of the individual. For many workers who fall into the everyday user profile (meaning they rely upon several communication tools such as email, instant messaging and the telephone) these workers often found themselves with too much in terms of buttons and functionality. The 9620 is designed specifically for the everyday user - who might make or receive only five to six calls per day. Providing just the right mix of functionality within a simple and intuitive interface.	Having a match between the communications needs of the everyday end user and the phone on their desk provides an office with great advantages. Allowing the user to confidently, proficiently perform their job, and make effective use of the right set of telephone features. Casual, everyday phone users have the base set of functionality in the 9620's smaller display with simple access to the basic features they need.

Additional Features

- Display – 4-level Grayscale with backlighting – 3.45" 320x160
- Handset – Hi-fidelity, wideband
- Speakerphone – Hi-fidelity
- Headset (additional) – Hi-fidelity, wideband
- Additional buttons and LEDs – 4 soft-keys: Applications Menu, Contacts, Call Log, Messages
- Line appearances / features (scrollable) – 12
- USB interface
- Adapter slots – 1
- Module interface – 1
- Supported # of 24-button modules – 0
- Ethernet switch (2 interfaces) – 10/100 – Gigabit module can be added
- PoE 802.3af – Class 2
- Adjustable display
- Wall mountable
- 2-position flip stand
- Customizable faceplate
- 12-month warranty
- G.722 wideband codec
- Hearing-aid, TDD compatible
- Mute and volume buttons
- WML applications
- 2 message-waiting lights
- Unicode support for 14 languages
- 4-way navigation key
- Call log, and Contacts applications
- Security, Authentication, QoS, H.323, and SIP compatibility

Feature	Benefit
Investment Protection - Modules and Adapters	Based on open standards with a modular platform, the Avaya 9640 allows enterprises to add a wide range of modules and adapters to further enhance employee productivity. This facilitates individual choice between Bluetooth, monaural (wideband or binaural) wideband headsets. A standard USB interface accommodates a range of USB devices. In the future, additional adapters and modules including those for Gigabit Ethernet and Bluetooth will be available. Investment protection and enhanced total cost of ownership are the key benefits. Traditionally, the only way to receive new telephone capabilities was to purchase a completely new phone. Designed with the future in mind, the Avaya 9640 provides a flexible approach for adding future functionality to current telephones.
9640 Designed just for the essential user	The 9640 is designed specifically for essential users--those who deem the telephone as critical for doing their jobs. The 9640 features a larger color display, with additional LED buttons and wideband audio support throughout. The 9640 supports a 24-button expansion module, which allows the user to access additional call appearances and feature keys. Finally, with one-touch access to mobility features, the 9640 user is reachable seamlessly. Matching the communication needs of the essential user with the phone on the user's desk gives a business great advantage--especially given the importance of the roles held by essential users (sales, customer-facing associates). The 9640 user can make confident, effective use of the right set of telephone features, which enables them to perform their jobs more proficiently and ultimately helps to satisfy customers and provide competitive advantages.

Technical Information

- The 9640 uses the g.722 codec open standard for wideband audio, which provides uncompromised sound quality.
- Software Requirements - Avaya Communication Manager Release 3.0 or greater
- Electrical Power - one-X Deskphone Edition telephones require electrical power, either from an 802.3af Power over Ethernet switch or a local power Avaya power supply. The 9640 is a Class 2 PoE device.

Key Benefits

- Productivity of Users – The productivity of end users is greatly enhanced through prompting for common telephony tasks, one-touch access to key features, and superior high fidelity audio.
- Richer Communication – The superior audio capabilities make conference calls and meetings more effective by requiring less reiteration. This has been found to reduce employee stress and fatigue.
- Investment Protection – Built on open standards with a modular platform that supports a wide range of modules and adapters to further enhance user productivity

Feature	Benefit
Intuitive User Interface	The intuitive, context-sensitive interface on the 9640 is designed to facilitate confident usage by featuring context-driven menus with on-screen prompts, enabling straightforward access to contact directory and call log. Essential users have full control of conference calls, including selective drop and mute. In addition, the 9640 provides a dedicated Call Forward/Mobility button for one-touch access to mobility features within Avaya Communication Manager. These include Extension to Cellular as well as Extend Current Call to cell phone. Easy access to common features such as Conference, Transfer, and Hold are now greatly enhanced with helpful prompts and tight integration with phone numbers in the contact list and call logs. The user interface on the 9640 provides easy access to critical Communication Manager mobility features, allowing users to be reached transparently whether at their desks or on the go.
Superior Audio Quality	The unique high-fidelity acoustics of the Avaya 9640, including wideband audio support in the speaker, handset, and headset deliver industry-leading audio that minimizes ambient noise. With the enhanced audio across high and low frequencies, it is easier for users to better understand others with different speech nuances or accents. For the essential end user, the person constantly on the telephone, the enhanced audio allows calls to be more productive by making team members better able to collaborate. On conference calls, users find it easier to distinguish and understand multiple speakers, aiding in collaboration and communications. Overall, communications are richer.
New design and color display	The 9640 features an improved, higher resolution display supporting 1/4 VGA color with backlighting. A four-way navigation button cluster is another new addition to the 9640, providing a familiar, cell phone-like interface for navigation and feature selections for the essential user. The new design facilitates better usage of the display and the built-in browser to improve access to information and use of telephone features.

9640 IP Telephone

A member of the Avaya one-X Deskphone Edition family, the 9640 IP Telephone with high resolution color display is specifically designed for the essential telephone user. Essential users are those who rely on the telephone as an essential tool for doing their jobs. Sales people, relationship managers, and attorneys are typical examples of essential users. The 9640 provides superior high fidelity audio, built-in "one touch" access to key Avaya Communication Manager mobility features, and a stylish and professional design.

The 9640 IP Telephone features a 3.8 inch (9.65 cm) diagonal high resolution color backlit display. The telephone supports up to 24 call appearances/administered feature keys with six concurrent line appearances visible at any time. The 9640 has several LED buttons throughout the front of the phone. Six LED line appearance buttons on the side of the display provide explicit status of different line appearances and administered features, while LEDs built into several buttons on the phone such as Mute, Message, and Headset provide an intuitive and simple experience for the everyday end user.

Avaya 9640 IP Telephone

Customizable color faceplates available

The user interface on the 9640 is helpful and intuitive. Completing call transfers and setting up ad hoc conference calls is simple and can be executed with confidence. The 9640 has a dedicated Call Forward/Mobility button, which provides direct access to Communication Manager Mobility features. Some of these features, such as Extension to Cellular and Extend Current Call, are critical to the essential user.

The 9640 supports a 24-button expansion module. This provides the essential user with additional call appearances, bridged appearances and administered feature keys including speed dials.

Avaya one-X® Deskphone 9670G

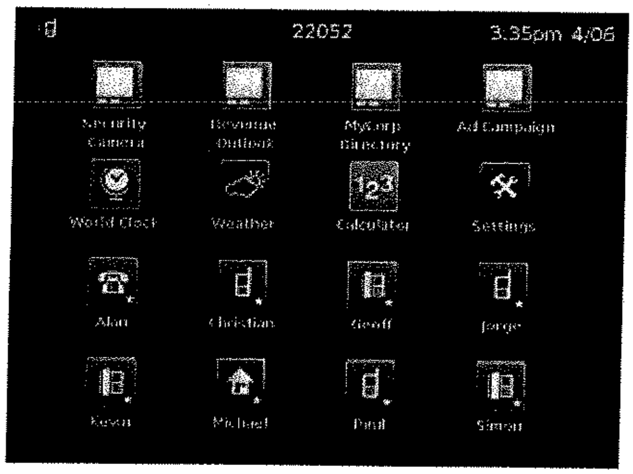
The Avaya one-X® Deskphone 9670G is designed for the "Essential" user: someone who is constantly on the phone, handling multiple calls and is often mobile. The latest addition to the Avaya one-X line – designed to deliver powerful and consistent communications across a variety of end user devices – the Avaya 9670G is rich in features, yet easy and intuitive to use.

Special Features of the 9670G:

- Big, bright, touchscreen interface with one- touch access to contacts, applications and real-time information (e.g., weather)
- Embedded "serverless" applications are responsive and don't require support from IT
- Integrated Gigabit Ethernet and Bluetooth support

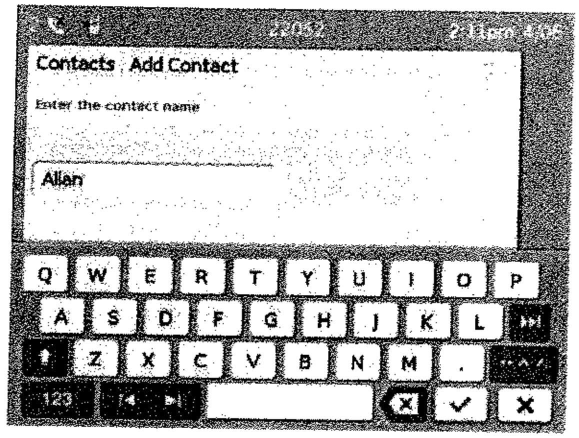
Appealing to your senses of touch, sight and sound, the 9670G touch screen is a feast for the eyes. Big and easy to see, this icon-based home screen is the command center of the phone, putting the user one touch away from dialing a contact, launching an application or accessing important information – there's even a virtual QWERTY keyboard for easy input and editing.

9670G Home Screen



The home screen's bright, bold icons eliminate the need for extraneous buttons, providing a user interface that is informative and visually pleasing. The audio experience delivered by the 9670G is every bit as remarkable. The wideband range (50-700Hz) provides rich, natural sounding low frequencies and crystal clear high frequencies – no more confusing "f" for "s" or wondering whose voice you're hearing on a conference call. The 9670G's industrial and acoustic design maximize the performance of the audio capabilities to deliver brilliant sound, whether on handset or headset or in hands-free mode.

9670G Virtual QWERTY Keyboard



Speech Enablement

The 9670G features speaker-independent speech recognition, there is no need to train the software. You can dial up to 250 contacts by simply speaking the person's name.

Embedded Always On Applications

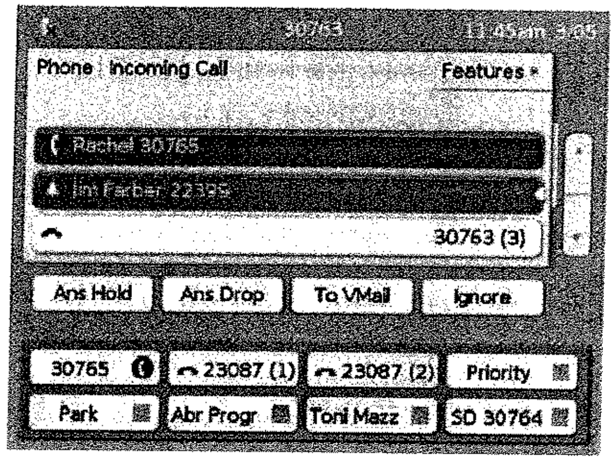
The 9670G leverages the "always on" nature of IP phones, so applications are always ready to be launched at the touch of a button. The screen provides quick access to embedded "serverless" Avaya applications (voice dialing, weather, world clock, and calculator); applications developed by Guam Community College or a third party; and favorites, up to eight one-touch speed dial icons. The result is a home screen that's highly functional and intuitive and requires little user training to make or forward calls, create conference calls, manage contacts, launch applications, access the call log and execute many other necessary tasks.

Gigabit Ethernet and Bluetooth

Integrated and ready to go, the 9670G comes equipped with Gigabit Ethernet and Bluetooth, streamlining the phone and simplifying operations and support for IT. The 9670G is able to provide these capabilities and still retain its green credentials since it is a PoE Class 2 device.

Link the deskphone and the mobile device. With the Avaya one-X series, the user has a common interface across all endpoint devices – deskphone, cell phone, computer and PDA. The 9670G makes it easy to shift a call from the deskphone to the mobile phone and vice versa – while the call is in progress – eliminating the inconvenience of having to call back in order to switch devices.

9670G Intuitive Interface



All that – and lower TCO too. The 9670G is designed for easy expansion and to minimize IT support needs. Upgradeable firmware delivers new capabilities with each new release, while USB and adapter interfaces also facilitate the addition of future applications. And PoE Class 2 power means no external power supply is needed – each phone costs less than US $6 per year to power, half the cost of a comparable PoE Class 3 device (vs PoE Class 2).

Specifications

- Firmware: S2.0 for 9670G; upgradeable
- Power Class: PoE Class 2
- Display: 5.1x3.8in / 13x9.75cm, full VGA; 640x480 pixels
- Voice codecs: G.722 G.711, G.729A/B, G.726A
- Communication protocol: H.323
- Adaptor Ports: 1
- USB Ports: 1
- Gigabit Ethernet: embedded
- Bluetooth: embedded
- Headset Interface: wideband
- QoS Support: 802.1 p/Q DiffServ

Requirements

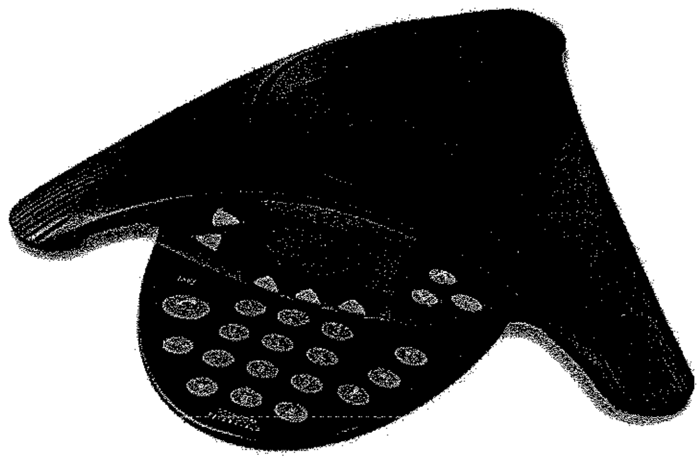
- Avaya Aura™ Communication Manager, Release 3.1 or higher.
- Local or Centralized Electrical Power — through POE 802.3af switch, or local power supply.

Avaya 1692 IP Speakerphone

A high-quality, IP-enabled speakerphone for conference rooms

The Avaya 1692 IP Speakerphone provides the convenience and productivity benefits inherent in a powerful, hands-free conference phone. It delivers the extensive set of Avaya Aura™ features directly to small, midsize and large conference rooms.

Avaya 1692 IP Speakerphone



Benefits

- Improved productivity during conference calls with hands-free full duplex operation delivering simultaneous two-way conversations. Reduced listener fatigue with high-fidelity audio from 220 Hz to 14,000 Hz, capturing both the deeper lows and higher frequencies of the human voice for conference calls that sound as natural as being there.
- Simplified wiring connects to your IP network with a 10/100 Base T Ethernet LAN connection. Simplified setup with integrated Power over Ethernet (PoE) with an AC power kit is available for non-PoE environments.
- Investment protection with easy upgrades via downloadable software and firmware.

Key Features

- Full Duplex Speakerphone with 360 degree, 12-foot microphone pickup. Automatic Gain Control intelligently adjusts microphone sensitivity based on where participants are seated in the conference room, making conversations clearer for all participants.
  - Microphone coverage expandable with two optional extension microphones
- RF Hardening technology resists interference from mobile phones and other wireless devices
- Room Coverage:
  - Up to 20 x 20 feet
    (without extension microphones)
  - Up to 20 x 30 feet
    (with extension microphones)

- High resolution backlit graphical display (255 x 128 pixels) enables robust call information and multi-language support
- 3 Context-Sensitive Soft Keys to give access to common telephony features
  - Automatically labeled from the system
- 5 Fixed Feature & Navigation Keys:
  On/Off Hook, Redial, Mute and Volume Up & Down
- 5 Menu and Navigation keys
- 12-key telephone keypad
- Single 10/100 Base T Ethernet connection
  - Full Duplex Ethernet connectivity with Auto-negotiation
  - 802.3 Flow Control
  - Supports VLAN
- G.711, G.729a, G722, Siren 14 Voice Codecs
- QoS Options of Diffserv and 802.1p/q
- Support for Simple Network Management Protocol (SNMP) version 2
- DHCP client and Statically (Manual) Configurable IP Addressing
- Downloadable Software for future upgrade capability with FTP/HTTPS server support in addition to HTTP and TFTP support
- Icon button labeling with English printing on the housing
- 5 Personalized Ring Patterns

Specifications

- 14.5" W x 12.25" L x 2.5" H
  (36.8 cm x 31.1 cm x 6.4 cm)
- Weight: 1.75 lb. (0.8 kg)
- Operating Temperature: 32° - 104° F
  (0° - 40° C)
- Universal power supply
  (100/240 V, 50/60 cycles)
- Dark Gray Color

Requirements

- Avaya Aura™ Communication Manager Release 4.0 or higher
- TN799C or higher circuit pack (C-LAN)
- TN2302AP circuit pack (Prowler)
  - Note: the Avaya S8300 Media Server does not require the two circuit packs listed above

Avaya OSPC Attendant, Operator and Information System

"OSPC takes switchboard operations to a higher level, enabling intelligent call routing with a personal touch"

Avaya Aura™ Communication Manager works directly with the Avaya OSPC solution to provide highly efficient telephony connections for attendants, receptionists and secretaries so they can quickly and easily provide communications and presence type information for any telephony connection request.

OSPC, which stands for *Operator Set for Personal Computers*, is a PC based software application that integrates telephony with external caller data and workforce information. This application can be easily expandable as Guam Community College's communication requirements evolve over time. As you know, it is easier to be successful if your customers and staff are satisfied with your telephone service. This includes the friendly and competent forwarding of incoming calls to the right contact. With OSPC, Avaya offers the capabilities your associates need to connect phone calls in a relaxed and friendly manner.

Expanding the Switchboard into an Intelligent Multi-media Assistant

With OSPC, an operator can support callers and the workforce with simplicity to do much more than merely extend a call. This solution enables simple switchboard operation with fast and direct call routing in ways that will soon be considered indispensable by company employees. For example, OSPC converts switchboard functions into a multimedia and text information system that enables operator access to a large range of information about customers and staff availability.

The interlinking of telephone data with staff information including absence notifications and customer backgrounds makes this solution highly productive for use at reception, switchboard and secretary desks. With this application, an operator can immediately see whether a staff member is in the building or not and who might be the right substitute if they are unavailable to take a call. This means that all callers can be connected to a competent person without delay.

First impressions count

Smooth customer relations are vital to business success. Often the first contact that customers have with Guam Community College is with your attendant or operator and they expect to be put through to the right person with minimum delay, preferably by a person with a smile in their voice. Thanks to the simple and convenient OSPC user interface, any attendant or operator can concentrate intelligently on customer needs and help them make the right connections efficiently.

With OSPC, your customers can be easily identified by name and linked with related contact information as the call is first connected. If desired, an attendant could welcome the caller by name or native language while connecting them to the right person; the responder who receives the customer call will know who is calling as well as have access to key information about the caller. The current business availability status for the responsible workforce contact for this customer or his substitute will be also displayed so the right connection can be made promptly.

OSPC grows with your business

Scaled to Guam Community College's present size, the application can subsequently be expanded with no difficulty as your business needs grow. The multi-location capability means that OSPC can be made available for simultaneous operation on a number of networked computers. Authorized and properly identified users can log on to the system from any computer equipped with OSPC and they will be greeted by a familiar user interface so they can start working right away from virtually any location.

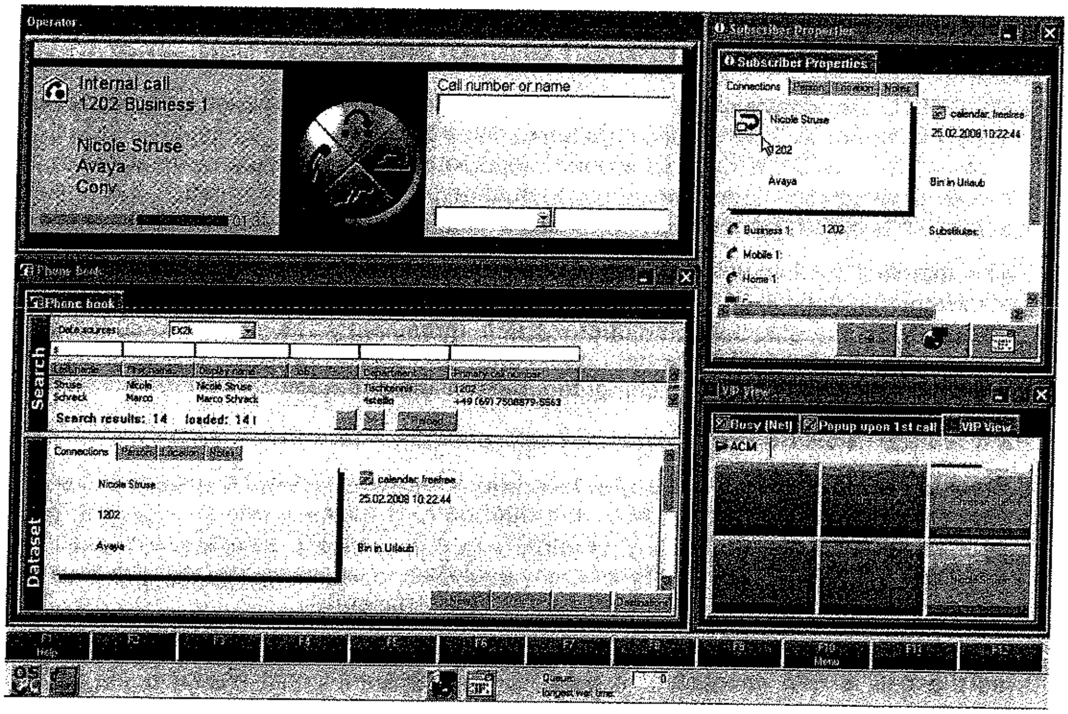
More than just call routing

OSPC software works with most standard PCs so the solution enables employees to perform call routing functions alongside other computing activities such as word processing or spreadsheet calculations. The solution supports efficient utilization of Guam Community College's human and hardware resources, allowing management to respond flexibly to any manpower situation arising at the workplace. The application can be operated like a classical switchboard or by using a keyboard or a mouse. A range of headsets is optionally available.

Individual workplace configuration

The application can be custom configured for every staff member and the individual components of the application can be displayed in separate windows arranged at will on the monitor screen. Where mouse operation is not feasible or desirable, the whole OSPC solution can be used with the keyboard only. Frequently required functions can be assigned to hotkeys on the keyboard or to free configurable buttons on OSPC.

The OSPC Interface — customizable to meet Guam Community College's specific needs

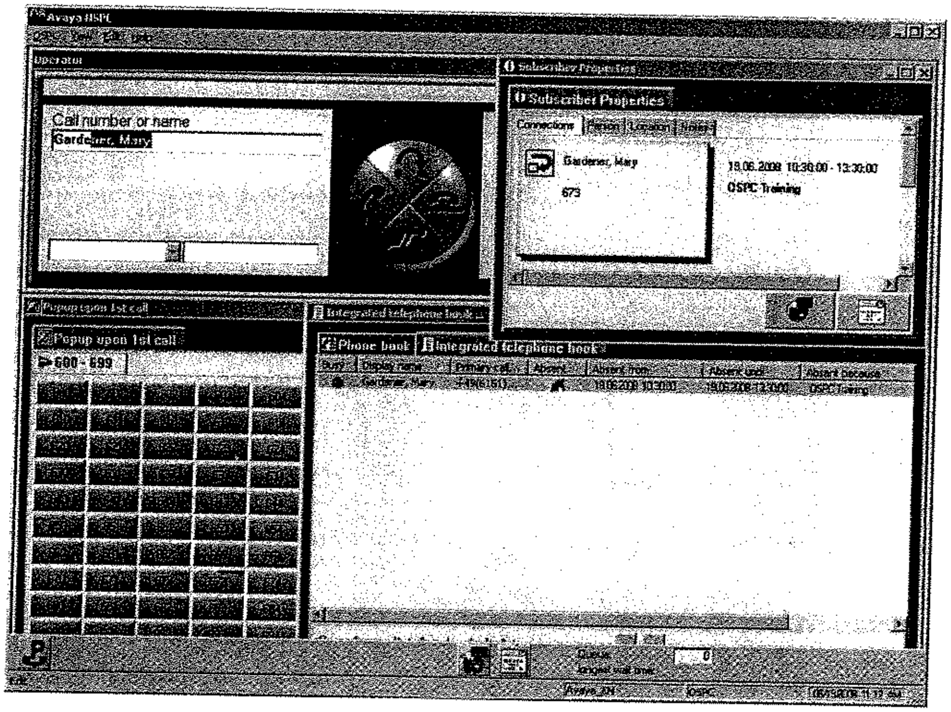


Intelligent connections support responsive interactions

Absences/presence information for all employees can be integrated with OSPC operations from Microsoft Outlook (Calendar and/or out of office assistant), IBM Lotus Notes (Calendar) and a web-based application. Key information from these applications is color displayed directly in the OSPC user interface. And with one click from this User Interface, an e-mail can be easily sent out for example to any Outlook or Lotus connected PC workstation so those who are not available by phone can also get an email about an interaction event.

OSPC also integrates the absence notifications from Outlook — Out of Office Assistant via the Exchange database connection so an "out of office" reply is automatically displayed before connecting the call to a person who is not there!

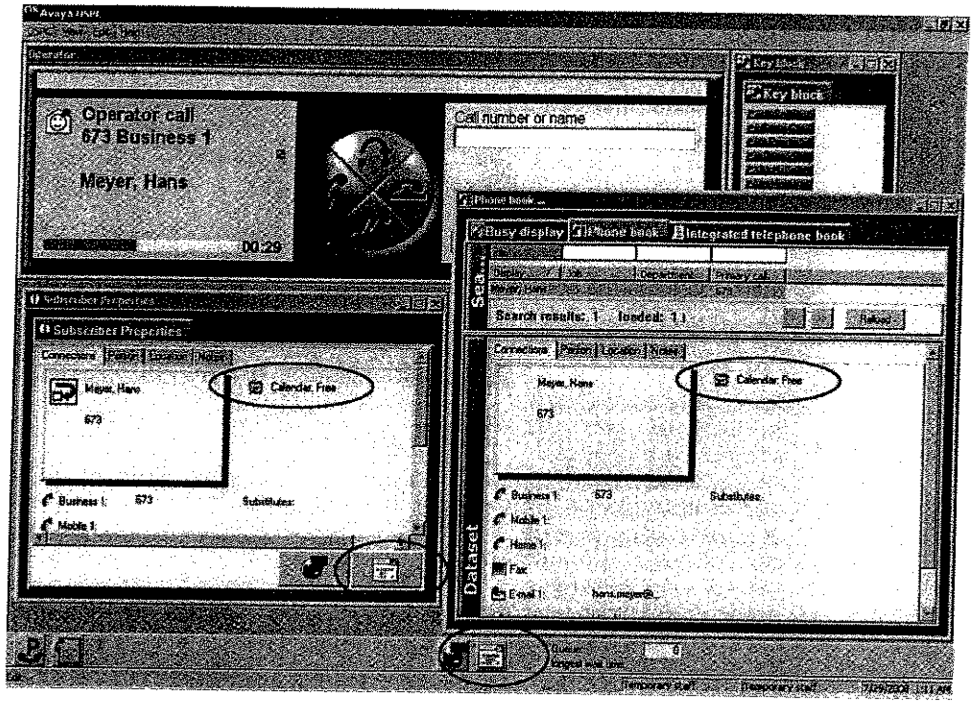
Integration with MS Outlook — Out of Office Assistant



Never lose track of things

Employees' calendar information from Outlook or Lotus Notes calendar appear in the color-coded status display and other dialog boxes which affords a quick overview at any given time.

OSPC Integrates with MS Outlook and Lotus Notes Calendar



Web-based absence notification

Employees who do not use Outlook or Notes can make use of a simple web-based tool to announce their absence from the office (e.g. lunch break, or meeting) and to indicate the duration of non-availability. Explanatory notes additionally entered by employees can be viewed via the phone book. Once the deadline expired, the display is reset to the default status and the employee is available again for all calls.

Busy lamp display for up to 2000 stations

The OSPC busy lamp fields (or blf) shows the status of the stations in blocks of 100 or 200 (max. 10 tabs) including number or name (where applicable) and both if available. The blf can be used for a call transfer and outgoing calls (destination keys) by clicking the button on the busy lamp field. The stations of connected gateways from far away locations will be also shown in the blf. Up to 2000 extensions can thus be monitored from a central point, allowing head office operators to put callers through to any desk at any branch office using convenient Drag & Drop techniques on their computer screens.

VIP treatment

The VIP View shows a designated subset list of the overall busy lamp field (blf) and important and/or frequently dialed extensions will typically be included in this list. The availability status function is not only supported on a local level but can also be extended to include remote locations. VIP view can show a selection of stations via the busy lamp display, and any tab section can be configured per department, location, workgroup, and more... For example, all stations from sales, some stations of service, parts of marketing might be displayed in a tab area as available for VIP connections.

Safety for sensitive data

Protecting access to sensitive business and personal data is an important priority today. Data integrity is an important issue particularly in companies where switchboard functions are performed by varying members of staff or part-time employees. OSPC enables the creation of user profiles which can be assigned for different levels of access authorization in line with the individual employees' areas of responsibility.

Enhanced efficiency thanks to task splitting

Task splitting is a very effective tool. For example, if Guam Community College needs two switchboard stations to handle incoming phone traffic during busy hours. One is manned by a longstanding employee, the other one by recently hired part-timer. OSPC allows two completely different user profiles to be established – one with extensive access rights to the customer and employee database, the other one exclusively for routing calls to given extensions. This flexible arrangement ensures optimum efficiency and effective data protection at the same time.

Central data maintenance

User data and profiles are stored in a central database. With user identification and access availability secured, the appropriate settings are enabled. Customer specific databases can also be connected via ODBC / LDAP interfaces with OSPC or to databases such as Active Directory System or Domino Server (Lotus Notes). This is a central data management/data care capability which reduces costs for system administration. The integrated statistics on the number of incoming calls and connected calls, waiting calls etc. provides valuable information on the utilization of attendants and improves resource planning. High availability for all locations can be maintained with local survivability system designs and via deployment of decentralized OSPC servers.

Accessibility Support for Disabled Workers

The OSPC system supports a Braille module from third party vendors which enables support for blind or visually impaired persons. This solution permits routing of calls without the need to see a display.

All information can be received via the Braille lines, while the simple structure and straightforward familiarization process rapidly produce high efficiency on the part of the visually impaired member of staff. The Braille module can also work with other programs such as word processors, thus facilitating additional employee capabilities to perform computer work while maintaining OSPC accessibility.

Client server system

The client server system capability of OSPC allows the simultaneous access from multiple clients with shared data in the network. Employees can logon each OSPC client PC in the enterprise and receive their customized user interface. All other functionality of office software suites and applications can be used in parallel to the typical attendant's work with mouse, keyboard and headset that uses the OSPC solution.

OSPC Requirements

- OSPC supports Communication Manager release 3.1 and higher
- OSPC Client or single solution (one PC for server/client)
- PC with 2 GHz, 1 GB RAM
- OS: Windows XP SP2 recommended
- 17" or larger TFT Monitor with 1280x1024 pixels
- OSPC Server
- PC with 2 GHz, 2GB RAM
- OS: Windows server 2003 / also on VMWare

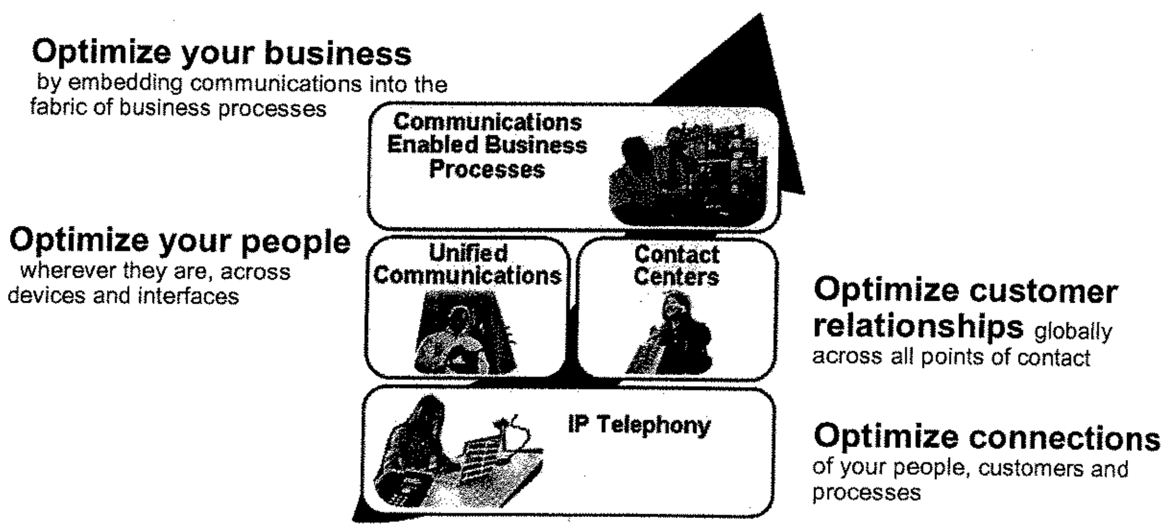
Avaya Call Center

Avaya Call Center provides the framework for a total customer service solution. Call Center is built upon proven and innovative automatic call distribution (ACD) technology. It offers a suite of call routing and resource selection capabilities designed to help Guam Community College's agents handle calls effectively and boost the overall level of your call center's productivity.

Avaya delivers intelligent communications allowing Guam Community College to identify changes; embrace the opportunity for change and turn that opportunity into a business reality. We do this by building on the foundation created with the deployment of IP telephony. Once the foundation is laid, we use the building blocks of unified communications and contact centers. By leveraging the advantages of the IP network, we can extend applications out to the edge with greater flexibility and lower Total Cost of Ownership. The next stage is the capstone of communication enabled business processes. By imbuing business workflows with the automation and intelligence of communications, Guam Community College can deliver on the promise of "the enterprise ready to serve" – creating a competitive advantage via superior customer service, to:

- Optimize business
- Optimize people
- Optimize connections
- Result – Optimize customer relationships, and real solutions to real business problems – today.

Avaya Intelligent Communication Vision



Capable of supporting up to 7,000 agents, the Avaya Call Center is well suited for companies like Guam Community College, with large advanced call centers as well as smaller less sophisticated centers. Guam Community College can rely on call center technology from Avaya to help support your success in today's increasingly competitive economy.

Avaya Call Center 5.0 offers three levels of software, plus optional features:

- Avaya Call Center Basic is included with every Avaya Aura™ Communication Manager, Release 5.0. It provides basic ACD features for a simple call center.
- Avaya Call Center Introductory provides all of the ACD functionality required to operate a small, basic call center, including basic conditional routing capabilities, for up to 50 agents at an extremely attractive per agent price.
- Avaya Call Center Elite is the most popular Call Center feature package. It includes Avaya Expert Agent Selection (EAS), which is the Avaya term for skills-based routing, and the full complement of advanced Call Vectoring (conditional routing) capabilities. The Elite package also includes features like Avaya Virtual Routing and IP Agent Shared Control. This package also licenses the use of the Avaya Agent Deskphone 16CC, a new SIP agent telephone. [The 16CC SIP Agent telephone requires Communication Manager, Release 5 and Avaya Aura™ SIP Enablement Services, Release 5].
- Business Advocate is a patented resource matching application available as an option to Call Center Elite.
- Advanced Segmentation is an application for segmenting your customer base using data-directed routing. It too, is available as an option with Call Center Elite.

Avaya Call Center Basic

Automatic Call Distribution (ACD)

Automatic Call Distribution (ACD) is an Avaya Call Center Software feature that resides onboard an Avaya Aura™ Communication Manager which is comprised of an Avaya Server and one or more Avaya Media Gateways. ACD is used to process high-volumes of incoming, outgoing, and internal calls and distributes them to groups of extensions called a hunt group or split. An "ACD split" is simply a hunt group that is designed to receive a high volume of similar inbound ACD calls. Calls to a specific split are automatically distributed among the staffed call takers known as agents, or hunt group members, assigned to that split. Inbound calls queue to the split until a staffed agent is available (idle). ACD allows a system administrator to create an efficient call management environment. This administrator can add or remove splits from the system, add or remove announcements, add or remove agents, add trunk groups and route calls to the appropriate splits. The administrator can also specify ACD measurement criteria to provide reports on ACD efficiency.

Avaya Call Center Basic includes basic ACD functionality such as call queuing, basic announcements, direct (linear) call distribution or uniform call distribution (most idle agent hunting), agent login/agent logout, support for the agent answering options of Automatic Answer and Manual Answer and support for staffed (logged-in) agents to be in any one of desktop telephone set controlled work modes [Auto-In, Manual-In, After Call Work (ACW) and Auxiliary (AUX) Work]. Additionally, support with Avaya Call Center includes the ability for staffed agents logged into multiple splits, Multiple Call Handling, Service Observing, VuStats, and more. Basic Call Management System is also provided with Call Center Basic at no additional charge and provides "built-in" Call Center reporting capabilities without the use of any adjunct server or software. As robust as all the above capabilities sound (they are recapped in our Avaya Call Center Software Overview document) Avaya does not envision many call centers actually running their day-to-day business off Avaya Call Center Basic. The idea behind offering Avaya Call Center Basic as standard with every Communication Manager is to enable a your firm to use rudimentary ACD capabilities in support of one or more help desks.

When a call arrives at a split, the ACD software checks to see if a staffed agent is available (idle) to handle the arriving inbound ACD call. If a staffed agent is not available (idle), or is already busy on a call then the call enters the split's queue. Calls queue only if there are no staffed (logged-in) agents available, and if the system-wide feature of Dynamic Queue Slot Allocation for Hunt Groups is in use. Alternatively the Communication Manager might not be administered with Dynamic Queue Slot Allocation for Hunt Groups if the administrator prefers to assign some fixed quantity (number) of queue slots to each skill (hunt group). In that case, if an ACD calls arrives for a split that has no logged-in (staffed) available agents and a queue is assigned to the split, and the queue is not full the call will queue in the split until answered. A split queue is a holding area for inbound ACD calls waiting to be answered. When a call is put into queue, the caller may hear one or more delay announcements, typically followed by music-on-delay, and/or silence, depending on the treatment that has been assigned for the split. Inbound ACD calls enter the queue at the bottom of the queue and move upwards over time to the top, or head, of the queue. After the queued call reaches the head of the split's queue, it connects to the next available (idle) agent for that split.

Agent Call Handling Using Avaya Call Center Basic

- Agent login and logout – To receive ACD calls, an agent must log into the Avaya Communication Manager. An agent can be logged into and therefore staffed multiple splits. If a hunt group is measured by an Avaya Call Management System or Basic Call Management System, an agent must enter a login ID.

- Agent Login – To log-in, an agent goes off-hook and dials the login feature access code (FAC), followed by the split number and the login ID, if required. A login button may be assigned on the agent's voice terminal. If login is successful, the agent automatically enters Auxiliary (AUX) Work mode for that split. The Auxiliary Work button lamp for that split, lights steadily and the agent hears the confirmation tone.
- Agent Logout – The agent should log out when they leave for an extended period of time and are therefore unavailable for receiving ACD calls. If an agent is logged into multiple splits, the agent should log out of each split. When temporarily unavailable for calls, an agent should use Auxiliary work mode, rather than logging out of the Communication Manager altogether. To log out of a split, an agent goes off-hook and dials the logout FAC followed by the split number. A logout button may be assigned on the agent's voice terminal. If logout is successful, the agent hears confirmation tone and work-mode button lamps darken.

- Agent answering options – An agent can answer ACD calls by using a headset, handset, or speakerphone. You can assign an agent as either Automatic Answer or Manual Answer.
  - Automatic Answer – An agent assigned to Automatic Answer hears zip-tone and connects directly to incoming calls without ringing. This is the most prevalent agent answering setting.
  - Manual Answer – An agent assigned to Manual Answer hears ringing, and then goes off-hook to answer the incoming call.
- ACD work modes –At any given time, a staffed (logged-in) agent can be in one of the following four unique work modes. The agent can change their work modes at any time. To enter any work mode, an agent presses the button on the set that they are using or dials the FAC for that mode, depending on what has been administered for the agent's use. If the agent has no active or held calls, the new work-mode's button lamp lights steadily. If the agent has active or held calls, the lamp flashes until all calls are dropped, then the new work mode's lamp lights steadily.
  - Auxiliary Work mode – An agent should enter Auxiliary (AUX) Work mode whenever taking a temporary break. This makes the agent unavailable for ACD calls and removes them from the most-idle-agent queue. Both Basic Call Management System and Call Management System can continue to track the staffed agent. When an agent logs into a split, they automatically enters AUX Work mode for that split.
  - Auto-In mode – In auto-in mode, the agent automatically becomes available for answering new ACD calls upon disconnecting from an ACD call. Agents in the auto-in mode typically prefer to use Automatic Answer with zip-tone. This is sometimes referred to as "call forcing."
  - Manual-In mode – In Manual-In mode, the agent automatically enters After Call Work (ACW) mode for the split upon disconnecting from an ACD call and is not available for any ACD calls. Another term for After Call Work is Post Call Processing or "call wrap-up." To become available for incoming ACD calls, the staffed agent must manually reenter either auto-in mode or manual-in mode. While many call centers will make little use of Manual-In for taking the majority of their ACD calls, it is good idea to use go ahead and administer Manual-In on the agent's set so that the agent can enter Manual-In for their next to last call they take prior to either logging out of their shift or leaving their position for a scheduled break or meal.
  - After Call Work mode – An agent should enter ACW mode when he or she needs to perform ACD-related activities, such as filling out a form as a result of an ACD call. The agent is unavailable for ACD calls to all splits while in ACW mode. The administration on

the Avaya Server that is running your Communication Manager and Call Center Software determines whether the agent remains in the Most Idle Agent (MIA) queue while they in After Call Work. When an agent is in the Manual-In mode and disconnects from an ACD call, they automatically enters the ACW mode. Although no longer available for ACD calls, the agent is still available for non-ACD calls.

Overflow for Call Center Basic

Calls can overflow from one split to another, to a specific station within the system, an attendant, an automated attendant, or a voice response or voice messaging system. Calls can be directed to any location that can be accessed using a valid number in the Communication Manager's dial plan. Calls may overflow to any destination on the ACD, regardless of Port Network or Media Gateway support with complete tracking by the management information system because the Avaya Server driving the Communication Manager supervises all endpoints universally.

In Call Center Basic, overflow is accomplished using the Call Coverage feature and is based on the number of rings. Overflow can be administered to consider up to three local destination paths. When an ACD call overflows from one split to a backup split queue and "priority on intraflow" is assigned, the intraflowed call has priority over all non-priority calls in the backup split queue.

Intraflow and Interflow for Call Center Basic

You can use Call Coverage with Intraflow to redirect ACD calls from one split to another conditionally, according to the coverage path's redirection criteria. For example, you can define a split's coverage path to automatically redirect incoming ACD calls to another split when a terminal is busy or unanswered. You can redirect calls to less busy splits, for more efficient call handling. You can use Call Forwarding with Intraflow to unconditionally forward calls for a split.

Interflow allows you to redirect ACD calls from a split on one communication server to a split on another communication server or external location. Use Call Forwarding All Calls with Interflow to unconditionally forward calls directed to a split to an off-premises location. Calls can be forwarded to destinations off the communication server (that is, phone numbers on the public telephone network).

Avaya Aura™ Communication Manager Messaging

Today's messaging requires more than just sending and receiving separate voice mail, email, and faxes. Today's tech savvy users require a multimedia messaging solution—one that can enable easy access to voice, fax, text, and file attachments from the user's choice of telephone or computer interface, with the ability to mix those media types within a single message. Avaya Aura Communication Manager Messaging provides this exact functionality for a small enterprise, as well as smaller locations of a larger enterprise. This architecture satisfies the short-term needs of a smaller location while supporting the evolutionary movement of messaging toward a more comprehensive unified communication solution.

Communication Manager Messaging is embedded application residing on the same server as Avaya Communication Manager. As such, it utilizes the same basic platform functionalities, combining many administrative duties into a single task. This integration occurs using H.323 and/or QSIG over an IP trunk. A cost-savings advantage is the license file generated for Avaya Communication Manager includes the license requirements for the messaging application. No additional licensing is required.

The number of ports and users is defined based on the Avaya Media Servers utilized, scaling from 12 to 250 ports, and 450 to 6,000 users.

Key Benefits

An Avaya Communication Manager Messaging solution provides the following key benefits:

- A telephone user interface and a graphical user interface, putting the choice of accessing messages via phone, computer, or wireless device from the office, at home, or on the road. In this way, users with different needs, preferences, and capabilities always remain connected.
- Integrated fax messaging combines the send and receive capabilities of a standalone fax machine or fax modem on a computer so that users receive incoming faxes in their mailboxes. In addition to listening to an attached voice comment from the telephone user interface, fax messages may be printed to a fax machine, a computer with a fax modem, a LAN printer, or a fax-enabled system such as another Communication Manager Messaging system. Using the graphical user interface, faxes may be viewed from the Inbox and stored there locally for convenient future reference.
- The INTUITY Message Manager application provides an easy-to-use graphical user interface. After entering their passwords, Message Manager provides users their entire mailbox contents at a glance, including voice, fax, and email messages, binary attachments, and system broadcasts. Users can select messages in any order to quickly locate and act upon based on their own priorities, without the need to move through all messages sequentially. On-screen controls are available to control playing, replying, forwarding, saving, deleting, undeleting, adding a hyperlink, spell-checking, adding a signature, recording and addressing messages to a variety of destinations with markings, editing the subject line, as well as managing greetings, passwords, and distribution lists, including viewing members. Playing and recording voice may be done either through the telephone or through the computer microphone and speakers.
- Using Message Manager, the user can sort and organize voice, fax, and using Internet Messaging, email messages into unlimited nesting personal folders on their computers, adding a text description for future reference, allowing for indefinite archiving. The messages saved in the Personal Folders can also be dragged and dropped back into the user's mailbox for re-sending or editing.
- Text-to-speech capabilities provide a voice rendering of the entire text component of a message received in Message Manager from any device, including a home phone, cellular phone, and a

computer. If a file attachment is included, it is not voiced; however, summary information is provided regarding the size of the attached file.

- Message Manager users can receive, forward, display, and print text messages when received from other versions of Avaya messaging systems via digital networking.
- Internet Messaging gives Message Manager and POP3 or IMAP4 client users the capability to send multimedia messages from the messaging software to any email address. The recipients can access these messages as multi-part MIME messages using commercially-available email software. Voice components are played with the G711 wave format through the PC.
- Whether it's multiple locations and/or customers on more than one continent or employees who need multilingual capabilities, it's important to standardize on a messaging platform that not only provides a multitude of languages, but also meets the stringent installation requirements in a number of countries. Two languages—American English and TTY/TDD for the hearing impaired—are pre-loaded from a superset of 35 available languages that may be requested and installed free of charge at installation.
- Callers can be given the option to select a language at the first Automated Attendant prompt and then hear subsequent prompts in that language. Using the Call Answer Language Choice feature, the user records two separate personal greetings, one for each language choice presented to callers. The caller interacts with the system in the language designated as that mailbox's primary language, with the option of switching to the secondary language. For users, as soon as a user on a multilingual system logs in, the system switches to the user's chosen login announcement set.

Standard Features

The following are standard unified messaging features of Modular Messaging:

- Automated Attendant and menus
- Outside caller features to include context-sensitive prompts, names directory, and message recording features
- Multilingual prompts available for menus, by callers' choices, and for users' mailbox greetings
- Prompts in the following languages: Arabic – Voices in both Male, Female; Chinese – Cantonese, Mandarin PRC, Mandarin Taiwan; Croatian; Czech Republic; Danish; Dutch; English – British Standard, TDD/TTY, United States, US Numbers; Finnish; French – Canadian, Parisian; German; Greek; Hindi; Hungarian; Indonesian; Italian; Japanese; Korean; Malay; Norwegian; Polish; Portuguese – Brazilian; Russian; Slovak; Spanish – Castilian, Latin America; Swedish; Thai, and Turkish.
- Mailbox parameters that include size of mailbox, length of messages, and automatic message deletion
- User greetings that include a standard, busy/no answer, internal/external, after hours
- Messaging addressing to single address and multiple addresses by extension or name, personal, and global distribution lists that can include fax and email addresses; broadcast messages
- Message delivery markings of urgent, private, future delivery
- Message notification at desktop, cell phone, offsite location, and pager
- Message review controls of new message count, envelope information, industry-standard options for playing messages, speed, volume, move forward/ backward

- End of message options such as save, delete, replay, reply, forward
- System diagnostics that transparently run 24 hour a day with multiple monitoring tools and logs; system error and alarm capabilities
- Standard system management reports with printing and exporting capabilities

Security

Communication Manager Messaging user mailbox security is addressed in the following manner:

- The user is free to set a password between the administrator-set minimum and 15-digit maximum length. Users are encouraged to change their passwords often and can do so at any time from any touchtone telephone.
- No one can access a user's password, as passwords are not displayed on any administration form. To allow for forgotten passwords, the System Administrator has the ability to set a new default password that the user must reset upon login.
- The Password Aging feature enhances system security by forcing users to change their passwords regularly. When a password is about to expire, the user hears an announcement immediately after logging in.
- Users are required to correctly enter their passwords to access their mailboxes through either the Message Manager graphical user interface or the telephone user interface. Users can also change their passwords from either interface. For greater security, Message Manager also enforces other password restrictions, such as the required periodic password changes set for that user's mailbox. Additionally, when Message Manager is minimized on the computer screen, the user can be required to re-enter the password in order to maximize the application.
- Mailbox lockout after a set number of invalid entry attempts.

Communication Manager Messaging has multiple system security measures in place, including:

- Restricted access to the computer through a series of logins. Each login has specific privileges and capabilities associated with it.
- The server requires a valid login ID and password be provided before accepting commands from the administration port.
- The system name is suppressed and only a ten-character system identification code at the log-in prompt is displayed.
- Passwords and unique identifiers for each machine preserve security in the network.
- Log-in sessions are automatically terminated after a pre-defined period of inactivity.
- An Administration History Log allows for the examination of recent administrative changes as part of security audits or root-cause analysis for problem resolution.
- Integrated toll-fraud restrictions

Specifications

Communication Manager Messaging is an embedded software application residing on the same server as Communication Manager. The number of ports and users varies by server, as shown in the following table.

System Capacities		
Avaya Server	Maximum Number of Ports	Maximum Number of Users
S8300 Server	12 Call Answer Ports	450 Users
S8400 Server	20 Call Answer Ports	900 Users
S8800 Server	48 Call Answer Ports	2,400 Users

Data Sheet

Cisco Unified Communications Manager Version 7.1

Cisco® Unified Communications Solutions unify voice, video, data, and mobile applications on fixed and mobile networks, enabling easy collaboration every time from any workspace.

Product Overview

Cisco Unified Communications Manager is the powerful call-processing component of the Cisco Unified Communications Solution. It is a scalable, distributable, and highly available enterprise IP telephony call-processing solution.

New with Cisco Unified Communications Manager Version 7.1

Feature Highlights and Benefits

As your needs evolve, Cisco Unified Communications Manager continues to evolve to meet those needs. Cisco Unified Communications Manager Version 7.1 aims to lower the total cost of ownership for organizations and improve the communications experience for end users as well as system administrators. Some of the important features of the recent release follow:

- **IPv6 support:** Cisco has added support for IPv6 in Version 7.1. We support a dual-stack deployment of IPv4 and IPv6, enabling a graceful migration for organizations moving from IPv4 networks to IPv6 networks over time.
- **Abbreviated dialing enhancements:** Now even when you are on the phone (off-hook), you can use simple, abbreviated dialing functions to more quickly conference or collaborate with others.
- **Q.SIG variant provisioning:** Version 7.1 provides the ability to configure either International Organization for Standardization (ISO) or European Computer Manufacturers Association (ECMA) variants on a gateway or trunk basis to a single Cisco Unified Communications Manager cluster. These variants greatly enhance the ability to migrate from multiple private-branch-exchange (PBX) environments to a single Cisco Unified Communications Manager system.
- **Increased scalability:** Version 7.1 increases support of:
  - Configured locations from 1000 to 2000
  - Configured regions from 1000 to 2000
  - Gateways and trunks from 1100 to 2100

Table 1 presents a complete list of new features included in Cisco Unified Communications Manager Version 7.1.

Table 1. New Features in Cisco Unified Communications Manager 7.1

IPv6
• IPv6 dual-stack support
New Telephony Features
• Abbreviated dialing enhancements
• Drop any conference party from computer telephony integration (CTI)-enabled endpoint
• Reverse callback
• Simultaneous ring time-of-day access list
PBX Interoperability and Migration

EXHIBIT 4

• Q.SIG variant provisioning
Directory
• Support for Open Lightweight Directory Access Protocol (LDAP)
Video
• Support for H.235 and H.239 secure and extended video channels
Simplified Administration and Upgrades
• New IP phone migration tool
• Cisco Data Migration Assistant (DMA) enhancements
Servers
• Support for next-generation Cisco MCS 7816, MCS 7825, and MCS 7828 Media Convergence Servers and software-only equivalents

Product Specifications

Platforms

- Cisco MCS 7800 Series Media Convergence Servers, including the Cisco MCS 7816, MCS 7825, MCS 7828, MCS 7835, and MCS 7845
- Selected third-party servers: For details, visit: http://www.cisco.com/go/swonly.
- Cisco Unified Communications Manager Business Edition: For details, visit: http://www.cisco.com/en/US/products/ps7273/index.html.

The appliance model provides a platform for call processing with the software preloaded on a Cisco MCS platform; the software is optionally available as a DVD kit for equivalent customer-provided servers. The appliance comes with a single firmware image that includes the underlying operating system as well as the Cisco Unified Communications Manager application. The appliance is accessed through a GUI, and a command-line interface (CLI) has been added to facilitate diagnostics and basic system management, such as the starting or stopping of services and rebooting of the appliance. No access to the underlying operating system is necessary. All system management activities — for example, disk space monitoring, system monitoring, and upgrades — are controlled through the GUI. Because onboard agents are no longer supported on the appliance, all Cisco Unified Communications Manager management interfaces are enhanced to allow tight integration with third-party applications.

Additionally, the Simple Network Management Protocol (SNMP) interface has added an overall syslog performance MIB. The serviceability interface has instrumented, appliance-specific counters. The programming interface has added the capability to run **insert**, **update**, and **delete** database commands. To further enhance security, Cisco Security Agent for Cisco Unified Communications Manager comes preloaded on the appliance.

Bundled Software

- Cisco Unified Communications Manager Version 7.1, a call-processing and call-control application, is included.
- The Cisco Unified Communications Manager Version 7.1 configuration database contains system and device configuration information, including the dial plan.
- Cisco Unified Communications Manager administration software is included.
- Cisco Unified Mobility service is included.
- The Cisco Unified Communications Manager CDR Analysis and Reporting Tool (CAR) provides reports for calls based on call detail records (CDRs) that include calls on a user basis, calls through gateways, simplified call quality, and a CDR search mechanism. The tool also provides limited database administration -- for example, deletion of records based on database size.
- The Cisco Unified Communications Manager Bulk Administration Tool (BAT) allows administrators to perform bulk insert, delete, and update operations for devices and users. The application was enhanced in Version 6

to provide export and import of database information, including calling search space, device pool, and Cisco Survivable Remote Site Telephony (SRST). Version 7.1 further enhances the solution by adding many other features, among them hunt and pilot lists, computer telephony integration (CTI) route groups, transformation patterns, presence groups, message waiting, and mobility information.

- The Cisco Unified Communications Manager Attendant Console application is no longer bundled with Cisco Unified Communications Manager. It is, however, supported in this release for customers who have upgraded from a previous version.
- The Cisco Unified Communications Manager Real-Time Monitoring Tool (RTMT) monitors real-time behavior of the components in a Cisco Unified Communications Manager cluster. Cisco Unified Communications Manager RTMT uses HTTP and TCP to monitor device status, system performance, device discovery, and CTI applications. It also provides trace and log file management capabilities, including download scheduling of all trace and log files, user-defined events in trace and log files, and real-time monitoring of trace and log files. Cisco Unified Communications Manager RTMT can send email and page alerts when problems are detected. It connects directly to devices by using HTTP for troubleshooting system problems.
- The Cisco Conference Bridge application provides software conference bridge resources that can be used by Cisco Unified Communications Manager.
- The Cisco Unified IP Phone Address Book Synchronizer allows you to synchronize Microsoft Outlook or Outlook Express address books with Cisco Personal Address Book. After installing and configuring Cisco Personal Address Book, you can access this feature from the Cisco Unified IP Phone Configuration website.
- The Cisco Unified Communications Manager Locale Installer provides user and network locales for Cisco Unified Communications Manager, adding support for languages other than English. Locales allow you to view translated text, receive country-specific phone tones, and receive Tool for Auto-Registered Phone Support (TAPS) prompts in a chosen language when working with supported interfaces. You can download this application from the Cisco website as needed.
- The Cisco Dialed Number Analyzer is a serviceability tool that analyzes the dialing plan for specific numbers.
- Cisco Unified Communications Manager Assistant provides call-routing and display capabilities required by busy administrative assistants and their managers in a business environment. By combining a PC-based console application and various soft keys and display panes on Cisco Unified IP Phones, Cisco Unified Communications Manager Assistant can offer you job-specific tools to more efficiently manage calls in this environment. This function is also available as an XML service on the phone.
- The Cisco Unified Communications Manager JTAPI plug-in is installed on all computers that host applications that interact with Cisco Unified Communications Manager with the Java Telephony API (JTAPI). JTAPI reference documentation and sample code are included.
- Cisco Telephony Service Provider contains the Cisco Telephony Application Programming Interface (TAPI) service provider (TSP) and the Cisco WAV drivers that TAPI applications use to make and receive calls on the Cisco Unified Communications system.

Session Initiation Protocol (SIP) support is available in Cisco Unified Communications Manager with support for line-side devices, including IETF RFC 3261-compliant devices available from Cisco and other manufacturers. Cisco SIP-compliant devices include the Cisco Unified IP Phone 7905G, 7912G, 7940G, and 7960G models. SIP is also available on the Cisco Unified IP Phone 7906G, 7911G, 7931G, 7941G, 7941G-GE, 7942G, 7945G, 7961G, 7961G-GE, 7962G, 7965G, 7970G, 7971G, and 7975G models, as well as on the Cisco Unified IP Phone Expansion Module 7914.

The SIP trunk interface is available and conforms to RFC 3261, allowing support of video calls over the SIP trunk and improving conferencing and application support experiences when used with Cisco Unity® and Cisco Unified MeetingPlace® solutions.

Call Admission Control (CAC) helps ensure that voice quality of service (QoS) is maintained across constricted WAN links, and it automatically diverts calls to alternate public-switched-telephone-network (PSTN) routes when WAN bandwidth is not available. A web interface to the configuration database allows remote device and system configuration. HTML-based online help is available for users and administrators.

Cisco Unified Communications Manager supports Resource Reservation Protocol (RSVP) agent capability. The RSVP agent on a Cisco router extends CAC capability beyond a hub-and-spoke topology within a cluster. Now a call can be routed directly between two locations without having to traverse the hub, allowing alternative network topologies and more efficient use of networks.

The Cisco Unified IP Phone 7931G initially supported in Cisco Unified Communications Manager 6.0 with Skinny Client Control Protocol (SCCP) is now optionally available with SIP. This phone provides functions that are commonly needed in the commercial and retail environments. It provides 24 lighted line keys and four interactive softkeys that guide you through call features and functions. In addition, it provides hard hold, redial, and transfer keys to facilitate simple and rapid call handling.

SNMP is available to manage Cisco Unified Communications Manager, allowing managers to set and report traps on conditions that could affect service and send them to remote monitoring systems.

System Capabilities Summary

- Alternate automatic routing (AAR)
- Attenuation and gain adjustment per device (phone and gateway)
- Audio message-waiting indicator (AMWI)
- Automated bandwidth selection
- Automatic route selection (ARS)
- AXL Simple Object Access Protocol (SOAP) application programming interface (API) with performance and real-time information
- Basic Rate Interface (BRI) endpoint support: Registers BRI endpoints as SCCP devices
- CAC: Intercluster and intracluster
- Call coverage
  - Forwarding based on internal and external calls
  - Forwarding out of a coverage path
  - Timer for maximum time in coverage path
  - Time of day
- Call display restrictions
- Call preservation -- redundancy and automated failover -- on call-processing failure
- Call recording
- Codec support for automated bandwidth selection: G.711 (mu-law and a-law), G.722, G.722.1, G.723.1, G.728, G.729A/B, Global System for Mobile-Enhanced Full Rate (GSM-EFR), Global System for Mobile-Full Rate (GSM-FR) iLBC (internet Low Bitrate Codec), wideband audio (proprietary 16-bit resolution; 16-kHz sampled audio), and Advanced Audio CODEC (AAC) for use with Cisco TelePresence devices
- Digit analysis and call treatment (digit string insertion, deletion, stripping, dial access codes, digit string translation, and dial pattern transformation) [[Deleting the asterisk here because it doesn't seem to refer to anything.]]
- Database resiliency to increase feature availability for the following:

- Extension mobility
- Call forward all
- Message-waiting indicator (MWI)
- Privacy
- Device mobility
- Do not disturb
- End-user and Application User Certificate Authority Proxy Function (CAPF) for CTI
- Monitoring
- Hunt groups

- Device mobility changes in the location-specific information when a device moves within the cluster
- Dial-plan partitioning
- Distributed call processing
  - Deployment of devices and applications across an IP network
  - Virtual clusters of up to eight Cisco Unified Communications Manager servers for scalability, redundancy, and load balancing
  - Maximum of 7500 Cisco Unified IP Phones per Cisco Unified Communications Manager server and 30,000 per server cluster (configuration dependent)
  - Maximum of 100,000 busy-hour call completions (BHCCs) per Cisco Unified Communications Manager server and 250,000 per server cluster (configuration dependent)
  - Intercluster scalability to more than 100 sites or clusters through H.323 gatekeeper
  - Intracluster feature and management transparency
- Divert calls to voicemail (iDivert)
- Fax over IP: G.711 pass-through and Cisco Fax Relay
- Forced authorization codes and client matter codes (account codes)
- H.323 interface to selected devices
- H.323 FastStart (inbound and outbound)
- Hotline and private-line automated ringdown (PLAR)
- Hunt groups: broadcast; circular; longest idle; and linear, login, and logout
- Interface to H.323 gatekeeper for scalability, CAC, and redundancy
- IPv4
- Language support for client-user interfaces (languages specified separately)
- Multilevel precedence and preemption (MLPP)
- Multilocation: Dial-plan partition
- Multiple ISDN Protocol support
- Multiple remote Cisco Unified Communications Manager platform administration and debug utilities
  - Prepackaged alerts, monitor views, and historical reports with RTMT
  - Real-time and historical application performance monitoring through operating system tools and SNMP
  - Monitored data-collection service
  - Remote terminal service for off-net system monitoring and alerting
  - Real-time event monitoring and presentation to common syslog

  - Trace setting and collection utility
  - Browse to onboard device statistics
  - Clusterwide trace-setting tool
  - Trace collection tool
- Multisite (cross-WAN) capability with intersite CAC
- Off-premises extension (OPX)
- Outbound call blocking
- Out-of-band dual-tone multifrequency (DTMF)
- Programmable line keys
- PSTN failover on route no availability: AAR
- Q.SIG
  - Alerting name specified in ISO 13868 as part of the Connected Name Identification Presentation (SS-CONP)
  - Basic call
  - ID services
  - General function procedures
  - Call back: ISO/IEC 13870: 2nd Edition, 2001-07 (completion of calls to busy subscriber [CCBS] and call completion on no reply [CCNR])
  - Call diversion: SS-CFB (busy), SS-CFNR (no answer), and SS-CFU (unconditional); service ISO/IEC 13872 and ISO/IEC 13873, first edition 1995 - Call diversion by forward switching and by rerouting
  - Call transfer by join
  - H.323 Annex M.1 (Q.SIG over H.323) - ITU recommendation for Annex M.1
  - Identification restriction (Calling Name Identification Restriction [CNIR] and Connected Line) Identification Restriction (COLR) and Connected Name Identification Restriction (CONR)
  - Loop prevention, diversion counter and reason, loop detection, diverted to number, diverting number, original called name and number, original diversion reason, and redirecting name
  - MWI
  - Path replacement ISO/IEC 13863: 2nd Ed. 1998, and ISO/IEC 13974: 2nd Ed. 1999
- Station through trunk (Media Gateway Control Protocol [MGCP] gateways)
  - JTAPI and TAPI applications enabled with automated failover and automatic update
  - Triple Cisco Unified Communications Manager redundancy per device (phones, gateway, and applications) with automated failover and recovery
  - Trunk groups
  - MGCP BRI support (ETSI BRI basic-net3 user-side only)
- Security
  - Secure conferencing is available to all members of the conference.
  - Configurable operation modes: Nonsecure or secure modes can be configured.
  - Device authentication: New model phones have an embedded X.509v3 certificate; a CAPF is used to install a locally significant certificate in the phones.

  - Data integrity: The Transport Layer Security (TLS) cipher NULL-SHA is supported; messages are appended with the SHA1 hash of the message to help ensure that they are not altered on the wire and can be trusted.
  - Cisco Unified Communications Manager offers secure HTTP support for Cisco Unified Communications Manager Administration, Cisco Unified Communications Manager Serviceability, Cisco Unified Communications Manager User Pages, and Cisco Unified Communications Manager CDR Analysis and Reporting Tool.
  - Privacy: Signaling and media are encrypted, including Cisco Unified IP Phone 7906G, 7911G, 7921G, 7940G, 7931G, 7941G, 7941G-GE, 7942G, 7945G, 7960G, 7961G, 7961G-GE, 7962G, 7965G, 7970G, 7971G, and 7975G models; Cisco Unified Survivable Remote Site Telephony; and MGCP gateways.
  - Secure Sockets Layer (SSL) for directory: Supported applications include Cisco Unified Communications Manager BAT, Cisco Unified Communications Manager CDR Analysis and Reporting Tool, Cisco Unified Communications Manager Admin User Pages, Cisco Unified Communications Manager Assistant Admin Pages, Cisco Unified IP Phone Options Pages, Cisco Conference Connection, Cisco CTI Manager, Cisco Communications Manager Extension Mobility, and Cisco Communications Manager Assistant.
  - A universal-serial-bus (USB) eToken containing a Cisco rooted X.509v3 certificate is used to generate a Certificate of Trust List (CTL) file for the phones and to configure the security mode of the cluster.
  - Phone security: Trivial File Transfer Protocol (TFTP) files (configuration and firmware loads) are signed with the self-signed certificate of the TFTP server; the Cisco Unified Communications Manager system administrator can disable HTTP and Telnet on IP phones.
  - SIP trunk (RFC 3261) and line side (RFC 3261-based services)
  - Cisco Unified SRST
- Shared resource and application management and configuration
  - Transcoder resource
  - Conference bridge resource
  - Topological association of shared resource devices (conference bridge, music-on-hold [MoH] sources, and transcoders)
  - Media termination point (MTP): Support for SIP trunk and RFC 2833
  - Annunciator
- Silence suppression and voice activity detection (VAD)
- Silent monitoring
- Simplified North American Numbering Plan (NANP) and non-NANP support
- SIP trunk Call Admission Control (SIP CAC)
- T.38 fax support (H.323, MGCP, and SIP)
- Third-party applications support
  - Broadcast paging: Through foreign exchange station (FXS)
  - Simple Messaging Desktop Interface (SMDI) for MWI
  - Hook-flash feature support on selected FXS gateways
  - TSP 2.1
  - JTAPI 2.0 service provider interface
  - Billing and call statistics
  - Configuration database API (Cisco AXL)

- Time-of-day, day-of-week, and day-of-year routing and restrictions
- Toll restriction: Dial-plan partition
- Toll-fraud prevention
  - Prevent trunk-to-trunk transfer
  - Drop conference call when originator hangs up
  - Require forced-authorization codes
- Unified device and system configuration
- Unified dial plan
- Video codecs: H.261, H.263, H.264, and Cisco Wideband Video Codec (Cisco Unified Video Advantage)
- Video telephony (SCCP, H.323, and SIP)

Summary of User Features

Note: Asterisks (*) in this list indicate SIP support for Cisco Unified Communications Manager 7.0.

- *Abbreviated dial
- *Answer and answer release
- *Auto answer and intercom
- *Callback busy and no reply to station
- *Call connection
- *Call coverage
- *Call forward: All (off net and on net), busy, no answer, no bandwidth, and not registered
- *Call hold and retrieve
- Call join
- *Call park and pickup
- *Call pickup group: Universal
- *Call pickup notification (audible or visual)
- *Call status per line (state, duration, and number)
- *Call waiting and retrieve (with configurable audible alerting)
- *Calling line identification (CLID) and calling party name identification (CNID)
- Calling line identification restriction (CLIR) call by call
- *Conference barge
- *Conference chaining
- *Conference list and drop any party (impromptu conference)
- *Dialed-number display
- *Direct inward dialing (DID) and direct outward dialing (DOD)
- *Directed call park with busy lamp field (BLF)
- *Directory dial from phone: Corporate and personal
- *Directories: Missed, placed, and received calls list stored on selected IP phones
- *Distinctive ring for on- and off-net status, per-line appearance, and per phone
- *Do not disturb (do not ring and call reject)
- *Drop last conference party (impromptu conferences)

- *Extension mobility support
- *Hands-free, full-duplex speakerphone
- *HTML help access from phone
- *Hold reversion
- *Immediate divert to voicemail
- *Intercom with whisper
- *Join across lines
- *Last-number redial (on and off net)
- *Log in and log out of hunt groups
- Malicious-call ID and trace
- *Manager-assistant service (Cisco Unified Communications Manager Assistant application) proxy line support
    - Manager features: Immediate divert or transfer, do not disturb, divert all calls, call intercept, call filtering on CLID, intercom, and speed dials
    - Assistant features: Intercom, immediate divert or transfer, divert all calls, and manager call handling through assistant console application
- *Manager-assistant service (Cisco Unified Communications Manager Assistant application) shared-line support
    - Manager features: Immediate divert or transfer, do not disturb, intercom, speed dials, barge, direct transfer, and join
    - Assistant features: Handle calls for managers; view manager status and calls; create speed dials for frequently used numbers; search for people in directory; handle calls on their own lines; immediate divert or transfer, intercom, barge, privacy, multiple calls per line, direct transfer, and join; send DTMF digits from console; and determine MWI status of manager phone
- *Manager-assistant service (Cisco Unified Communications Manager Assistant application) system capabilities: Multiple managers per assistant (up to 33 lines) and redundant service
- *Manager-assistant service
- *MWI (visual and audio)
- *Multiparty conference: Impromptu with add-on meet-me features
- *Multiple calls per line appearance
- *Multiple line appearances per phone
- *MoH
- *Mute capability from speakerphone and handset
- *On-hook dialing
- *Original calling party information on transfer from voicemail
- *Privacy
- *Real-time QoS statistics through HTTP browser to phone
- *Recent dial list: Calls to phone, calls from phone, autodial, and edit dial
- *Service URL: Single-button access to IP phone service
- *Single-button barge
- *Single-directory number and multiple phones: Bridged line appearances
- *Speed dial: Multiple speed dials per phone

- *Station volume controls (audio and ringer)
- *Transfer: Blind, consultative, and direct transfer of two parties on a line
- *User-configured speed dial and call forward through web access
- *Video (SCCP, H.323, and SIP)
- *Web services access from phone
- *Web dialer: Click to dial
- *Wideband audio codec support: Proprietary 16-bit resolution, 16-kHz sampling rate codec

Cisco Unified Mobility

The Cisco Unified Mobility service helps mobile workers direct their inbound business calls to their IP phone number and initiate outbound business calls as if they were at their Cisco Unified IP phone - all from the mobile phone (or other remote phone destination). They can answer incoming calls on the desk phone or mobile phone, pick up calls between the desk phone and mobile phone without losing the connection, and originate enterprise calls from a mobile or other remote phone. Cisco Unified Mobility is included in Cisco Unified Communications Manager 7.1 and provides the following features:

- Allowed and blocked call filters
- Caller identification
- Call screening and call divert
- Call tracing
- Cisco Mobile Voice Access
- Desktop pickup
- Directed call park through DTMF
- Mobile call pickup
- New mobility device model type
- Remote on and off control
- Reverse callback to nonmobile number
- Security and privacy for Cisco Unified Mobility calls
- Single enterprise voice mailbox
- Simultaneous desktop ringing
- System administrator-controllable user profile access
- Voice-based access with user identification and personal identification number protection

Summary of Administrative Features

- Application discovery and registration to SNMP manager
- AXL SOAP API with performance and real-time information
- Cisco Unified Communications Manager BAT (including new import and export capabilities)
- CDRs
- Cisco Unified Communications Manager CDR Analysis and Reporting Tool
- Call forward reason code delivery
- Centralized, replicated configuration database and distributed web-based management reports
- Configurable and default ringer WAV files per phone

- Configurable call forward display
- Database automated change notification
- Date and time display format configurable per phone
- Debug information to common syslog file
- Device addition through wizards
- Device-downloadable feature upgrades: Phones, hardware transcoder resource, hardware conference bridge resource, and voice-over-IP (VoIP) gateway resource
- Device groups and pools for large-system management
- Device mapping tool: IP address to MAC address
- Dynamic Host Configuration Protocol (DHCP) block IP assignment: Phones and gateways
- Dialed Number Analyzer (DNA)
- Dialed-number translation table (inbound and outbound translation)
- Dialed-number identification service (DNIS)
- Enhanced 911 service
- H.323-compliant interface to H.323 clients, gateways, and gatekeepers
- JTAPI 2.0 CTI
- LDAP Version 3 directory interface to selected vendors' LDAP directories: Active Directory and Netscape Directory Server
- MGCP signaling and control to selected Cisco VoIP gateways
- Native supplementary services support to Cisco H.323 gateways
- Paperless phone DNIS: Display-directed button labels on phones
- Performance-monitoring SNMP statistics from applications to SNMP manager or to operating system performance monitor
- QoS statistics recorded per call
- Redirected DNIS (RDNIS) inbound and outbound (to H.323 devices)
- Select specified line appearance to ring
- Ability to select specified phone to ring
- Single CDR per cluster
- Single-point system and device configuration
- Sortable component inventory list by device, user, or line
- System event reporting to common syslog or operating system event viewer
- TAPI 2.1 CTI
- Time zone-configurable per phone
- Cisco Unity software user integration
- TAPS
- XML API for IP phones
- Zero-cost automated phone moves
- Zero-cost phone adds
- Data migration assistant
- Log partition monitor

- Disaster recovery framework
- Cisco Security Agent for Cisco Unified Communications Manager
- IP Security (IPsec) and certificate management
- CDR delivery manager
- Command-line interface
- Enhanced remote access through serial, console, and Secure Shell (SSH) Protocol
- Scheduled provisioning with Cisco Unified Communications Manager BAT
- Scheduled trace collection
- User-defined events
- Real-time trace monitoring
- Enhanced upgrade process to minimize service downtime
- Enhanced installation process to minimize install time
- Installation answer file for no-touch installation
- Syslog to SNMP trap MIB
- Enhanced AXL SOAP API to modify the database

SIP Trunk and Endpoint Support

SIP trunk and endpoint support provides enhancements to support SIP and host SIP phones, improving interoperability and opening ways to develop innovative applications. Cisco Unified Communications Manager supports coexistence of SCCP and SIP phones, allowing migration to SIP while protecting investments in existing devices. Cisco Unified Communications Manager includes the following major SIP functions:

- Native support of SIP devices
- CTI for Internet service provider (ISP) phones
- Presence information for SIP devices, including support for PUBLISH
- Fault, configuration, accounting, performance, and security (FCAPS) enhancements to support SIP
- SIP trunk enhancements for external applications, such as conferencing and presence
- Third-party SIP devices supporting RFC 3261
- SIP line-side RFCs: RFCs 3261, 3262, 3264, 3265, 3311, 3515, and 3842
- SIP trunk RFC support: RFCs 2833, 2976, 3261, 3262, 3264, 3265, 3311, 3323, 3325, 3515, 3842, 3856, and 3891

Licensing

- Application and phone software licenses are enforced. The system manages the maximum number of devices that can be provisioned.
- Each device (Cisco Unified IP Phones, soft phones, third-party devices, and video devices) provisioned in the system corresponds to a number of device license units (DLUs), depending on its capabilities. The total number of units is managed in Cisco Unified Communications Manager to determine capacity.
- DLUs must be purchased to cover the number of devices connected to Cisco Unified Communications Manager.
- Third-party SIP devices require DLUs for operation with Cisco Unified Communications Manager.

Cisco Unified Workspace Licensing

This product is a part of Cisco Unified Workspace Licensing. Please visit http://www.cisco.com/go/workspace_licensing for more information and to determine whether Cisco Unified Workspace Licensing is appropriate for your customer.

Localization

The following user locales (languages) are supported:

Arabic, Bulgarian, Catalan, Chinese (Hong Kong), Chinese (simplified), Chinese (traditional), Croatian, Czechoslovakian, Danish, Dutch, Estonian, Finnish, French, French (Canadian), German, Greek, Hebrew, Hungarian, Italian, Japanese, Korean, Latvian, Lithuanian, Norwegian, Polish, Portuguese, Portuguese (Brazilian), Romanian, Russian, Serbian , Slovak, Slovenian, Spanish, Spanish (Latin American), Swedish, Thai, and Turkish.

The following network locales (tones and cadences) are supported: Argentina, Australia, Austria, Belgium, Brazil, Canada, China, Colombia, Cyprus, Czech Republic, Denmark, Egypt, Finland, France, Germany, Ghana, Greece, Hong Kong, Hungary, Iceland, India, Indonesia, Ireland, Israel, Italy, Japan, Jordan, Kenya, Korea Republic, Lebanon, Luxembourg, Malaysia, Mexico, Nepal, Netherlands, New Zealand, Nigeria, Norway, Pakistan, Panama, Peru, Philippines, Poland, Portugal, Russian Federation, Saudi Arabia, Singapore, Slovakia, Slovenia, South Africa, Spain, Sweden, Switzerland, Taiwan, Thailand, Turkey, United Kingdom, United States, Venezuela, and Zimbabwe.

Ordering Information

Software Upgrades

Cisco Unified Communications Manager 7.1 installation CDs and DVDs can be ordered for existing systems.

Customers with a Cisco Unified Communications Software Subscription running Cisco Unified Communications Manager 4.1 to 7.0 who want to upgrade to Cisco Unified Communications Manager 7.1 can order upgrades using the Product Upgrade Tool located at: http://www.cisco.com/upgrade.

If you are planning an upgrade to Cisco Unified Communications Manager Version 7.1, please refer to the upgrade program for supported servers at: http://www.cisco.com/go/swonly.

Hard disk capacity of 72 GB or greater and 2 GB of RAM are required.

Cisco Unified Communications Services

Cisco Unified Communications Services allows you to accelerate cost savings and productivity gains associated with deploying a secure, resilient Cisco Unified Communications Solution. Delivered by Cisco and our certified partners, our portfolio of services is based on proven methodologies for unifying voice, video, data, and mobile applications on fixed and mobile networks. Our unique lifecycle approach to services can enhance your technology experience to accelerate true business advantage.

For More Information

For more information about Cisco Unified Communications Manager, please visit http://www.cisco.com/go/unifiedcm or contact your local account representative.

Americas Headquarters
Cisco Systems, Inc.
San Jose, CA

Asia Pacific Headquarters
Cisco Systems (USA) Pte. Ltd.
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Europe Headquarters
Cisco Systems International BV
Amsterdam, The Netherlands

Cisco has more than 200 offices worldwide. Addresses, phone numbers, and fax numbers are listed on the Cisco Website at www.cisco.com/go/offices.

Printed in USA

C78-532158-01 06/09

PROFESSIONAL SERVICES

GTA included in this proposal a number of professional services that are an important part to have all components ofthe proposed solution work with the quality levels required to grow a profitable business out of the proposed end-user services. The solutions that GTA provides in this area are comprised of a matrix of services and customized applications.

Aspart of our proposal, GTA will deliver the following professional services for GCC:

a) Project Management
b) Installation
c) Integration, End-to-End Test Activities & Solution Validation
d) User Acceptance Test
e) Training
f) Launch Support & Operational Hand Over
g) Technical Support & Maintenance

PROJECT MANAGEMENT

GTA will build the project framework for the project deployment. The framework is based on our PC3M process "Project Collaboration, Control, and Communication Method". PC3M is deeply rooted in the principles outlined in Project Management Institute's (PMI's) Project Management Body of Knowledge (PMBOK), but takes this a step further and focuses the project team specifically upon the goals and objectives necessary for the successful completion of high end telecommunications projects. The assigned project manager tracks and coordinates resources, understands and assists the client in identifying project priorities communicating them to the team with controls, checklists, meeting agendas, status reports and other collaboration tools designed to sort out the myriad of conflicts that arise in a complex technical implementation. Using PC3M, the Project Manager communicates and manages priorities building consensus and focus with all elements of the project team, both internal and external.

Our management and technical work processes were developed to confront risks that might impede our ability to complete high quality work on time and within budget. We overcome risk-related obstacles through the application of our risk management processes. This is accomplished with a qualitative and quantitative risk analysis to identify appropriate indicators and measured responses to mitigate any potential issues. Additionally, we utilize a series of project-oriented Gantt Chart tools, Critical Path calculations in our scheduling process to quickly identify resource conflicts, interrelated delays and the impact of changes concerning key components and personnel. Microsoft Project is our tool of choice for our high level tracking tool. In-Progress Reviews (IPRs) will include an evaluation of technical progress, progress on milestones, draft deliverables, and resolution of any issues that arise between IPRs.

The Project Manager will ensure the project objectives are met in due time and budget and in line with the expected quality and result.

EXHIBIT 5

Major Responsibilities of Project Management:

- Definition of and consensus on detailed project scope
- Define solution and all components for the System
- Project change management including all related third party interactions, i.e. software and hardware component suppliers, inside wiring contractors, etc.
- Definition, communication, and management of deployment project plan(s)
- Project coordination and synchronization of implementation activities
- Project reporting and communications
- Project risk management and risk mitigation
- Master Project plan/timeline in Microsoft Project format
- Third-Party Management which include Meeting Planning, Escalation Process Compliance and Supplier Ownership of all third parties.

QUALITY CONTROL

Quality control is seamlessly incorporated into all of our planning and engineering processes to ensure the consistent level of professional services we offer all of our customers. Upon award of a contract, the project manager identifies key areas where quality can be measured and then defines key metrics to measure and control our quality processes. These quality metrics are communicated in our In-Progress Reviews (IPRs), and enable the project manager to quickly identify and mitigate any quality issues before they compound and adversely impact the project deliverables.

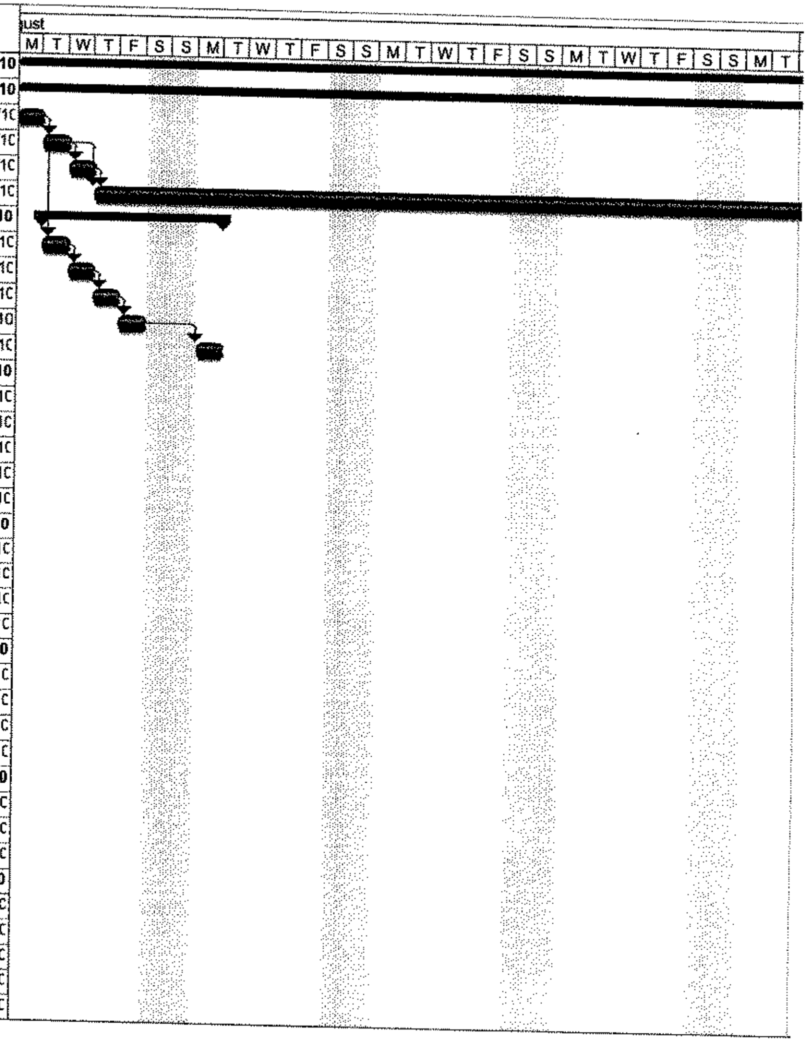
LEAD TIMES

The indicative lead time for delivery of the equipment on site is 4-6 weeks from time of purchase. The indicative overall project planning until user acceptance is estimated at 2 months from time of purchase. Project Plan is attached on next page with assumptions.

PROJECT PLAN

The following is a Gantt chart illustrating the activities involved in the installation, configuration and training of the new GCC VoIP Telephony System. We have identified a hypothetical completion date by assuming an award date of August 8, 2010:

ID	Task Name	Duration	Start	Finish
1	**Contract Issued & Notice To Proceed**	**49 days?**	**Mon 8/2/10**	**Thu 10/7/10**
2	**Requirements Analysis**	**33 days?**	**Mon 8/2/10**	**Wed 9/15/10**
3	Kickoff Meeting	1 day?	Mon 8/2/1C	Mon 8/2/1C
4	Verify Equipment Order	1 day?	Tue 8/3/1C	Tue 8/3/1C
5	Verify Telco Order Placed	1 day?	Wed 8/4/1C	Wed 8/4/1C
6	Receive & Inventory Avaya Equipment	30 days?	Thu 8/5/1C	Wed 9/15/1C
7	**Design**	**5 days?**	**Tue 8/3/10**	**Mon 8/9/10**
8	Review And Design IP Addressing	1 day?	Tue 8/3/1C	Tue 8/3/1C
9	Review and Design VLAN and Routing Plans	1 day?	Wed 8/4/1C	Wed 8/4/1C
10	Submit Ip Addressing Changes To Customer For Approval	1 day?	Thu 8/5/1C	Thu 8/5/1C
11	Review Communication Handling Requirements	1 day?	Fri 8/6/10	Fri 8/6/10
12	Develop, Document and Verify Dial Plan	1 day?	Mon 8/9/1C	Mon 8/9/1C
13	**Program/Build**	**5 days?**	**Thu 9/16/10**	**Wed 9/22/10**
14	Burn-In Avaya Equipment	1 day?	Thu 9/16/1C	Thu 9/16/1C
15	Install Gateway	1 day?	Fri 9/17/1C	Fri 9/17/1C
16	General Site Preparation - Verify Environmentals	1 day?	Mon 9/20/1C	Mon 9/20/1C
17	Verify Data Infrastructure Op - Troubleshoot	1 day?	Tue 9/21/1C	Tue 9/21/1C
18	Setup & Test Communication Mgr And Aura	1 day?	Wed 9/22/1C	Wed 9/22/1C
19	**Communication Manager Configuration & Testing**	**4 days?**	**Thu 9/23/10**	**Tue 9/28/10**
20	Build Communication Manager & Database	1 day?	Thu 9/23/1C	Thu 9/23/1C
21	Program Ext and Feature Sets	1 day?	Fri 9/24/1C	Fri 9/24/1C
22	Program Voice Applications	1 day?	Mon 9/27/1C	Mon 9/27/1C
23	Program Aura Voicemail	1 day?	Tue 9/28/1C	Tue 9/28/1C
24	**Installation And Testing**	**3 days?**	**Thu 9/23/10**	**Mon 9/27/10**
25	Install Communication Manager & Aura	1 day?	Thu 9/23/1C	Thu 9/23/1C
26	Telco Due Date For All New Trunks	0 days	Thu 9/23/1C	Thu 9/23/1C
27	Test Communication Manager	1 day?	Fri 9/24/1C	Fri 9/24/1C
28	Test PRI/ISDN circuits	1 day?	Mon 9/27/1C	Mon 9/27/1C
29	**Training**	**3 days?**	**Tue 9/28/10**	**Thu 9/30/10**
30	Prepare/Print GCC Custom Training Doc's	1 day	Tue 9/28/1C	Tue 9/28/1C
31	Initial End User Training	1 day?	Wed 9/29/1C	Wed 9/29/1C
32	Set & Detail Phones	1 day?	Thu 9/30/1C	Thu 9/30/1C
33	**GCC Cutover**	**3 days?**	**Fri 10/1/10**	**Tue 10/5/10**
34	Support And Maintenance	1 day?	Fri 10/1/1C	Fri 10/1/1C
35	Postcut Onsite Support Day1	1 day?	Mon 10/4/1C	Mon 10/4/1C
36	System Admin Training Review On Comm Mgr & Aura	1 day?	Tue 10/5/1C	Tue 10/5/1C
37	Final Review of Acceptance Testing Reports	1 day?	Wed 10/6/1C	Wed 10/6/1C
38	Final Implementation Invoicing	1 day?	Thu 10/7/1C	Thu 10/7/1C



INSTALLATION

The objective is to provide a successful installation of the proposed solution. GTA is responsible for the installation and commissioning of:

- Avaya Communication Manager
- Aura Server
- Media Gateway
- Instruments
- Power over Ethernet (POE) Switches

Based on the assumptions made regarding solution components, GTA has developed an estimated effort and determined the resources necessary to implement and commission the various components into GCC's network. The GTA VoIP Team proposes to perform a comprehensive site survey for the purpose of establishing a detailed understanding of the installation environment at the various sites where equipment is to be installed.

After each site is surveyed, the GTA VoIP Team will provide a site plan that describes the site specific installation plan that includes location of equipment to be installed. Drawings and digital photos, when appropriate, will be used to provide installation plan details and to memorialize the pre and post installation environment. Additionally, the installation plan will address specific plans for assuring that installation is accomplish in a manner that is astatically pleasing. The installation plan will address any site preparation that will be required as a result of equipment installation. The ultimate goal of the installation plan is to describe the installation process that will be used to install the VoIP equipment in an astatically pleasing fashion and any activity required to restore the site after installation is has been completed.

The plan will be provided to GCC's designated technical authority for approval prior to beginning installation and construction.

The installation activities to support VoIP will include installation of trunk lines, switching and server equipment, and a variety of terminal and premises equipment. The intent of the installation plan is to perform all of these tasks in a parallel to reduce the overall program installation interval. When the trunk lines are ready; the switches and servers will be ready; the terminal equipment will be ready and we will commence configuration and integration, test and turn-up of the system. We will focus on the bringing users on line in as the infrastructure becomes available. We will concentrate our work to provide service on a building by building or area by area basis.

Integration, End-to-End Test Activities & Solution validation

GTA TeleGuam will perform ongoing integration, testing and solution validation of the solution components.

- System Delivery
- Installation of Systems - software and associated hardware
- Configure and test System, ready for customer acceptance

User Acceptance Test (UAT)

The goal of UAT, is to test if the solutions deployed is working in GCC's environment according to GCC requirements and to achieve acceptances. GTA will support GCC during the UAT testing.

Training

GTA TeleGuam will provide the following training services associated with the system. The intent is to provide knowledge transfer to GCC staff on the system that has been installed. The services and activities:

- Formal classroom training on introduction and operation of individual components
- On-the-job training, with Integrator experts providing knowledge transfer to GCC staff on the operation of the solution, first customer contact maintenance, and administering of the systems.

Launch Support & Operational Handover

In order to guarantee a successful start of the new service, GTA TeleGuam will support GCC, ensuring a smooth handover of responsibilities and knowledge at the end of the project. Such assistance will make sure that experienced personnel perform all daily tasks. During the handover GTA TeleGuam shall provide relevant documentation including handover document and will conduct a handover session in which all still open issues will be discussed

Technical Support and Maintenance Services

The support services shall include:

a) Tier 1 support – Will be provided by GCC IT staff.
b) Remote Tier 2 helpdesk for technical support service by GTA TeleGuam
c) Minor software updates/ patches with bug fixes.
d) Technical analysis and resolution of problems, escalated by lower support tier.
e) Technical consultation for questions and queries related to the supported system.
f) Interface with the suppliers of third party embedded in the system. (when applicable)

EXHIBIT 6

PROFESSIONAL SERVICES

GTA included in this proposal a number of professional services that are an important part to have all components of the proposed solution work with the quality levels required to grow a profitable business out of the proposed end-user services. The solutions that GTA provides in this area are comprised of a matrix of services and customized applications.

As part of our proposal, GTA will deliver the following professional services for GCC:

a) Project Management
b) Installation
c) Integration, End-2-End Test Activities & Solution Validation
d) User Acceptance Test
e) Training
f) Launch Support & Operational Hand Over
g) Technical Support & Maintenance

PROJECT MANAGEMENT

GTA will build the project framework for the project deployment. The framework is based on our PC3M process "Project Collaboration, Control, and Communication Method". PC3M is deeply rooted in the principles outlined in Project Management Institute's (PMI's) Project Management Body of Knowledge (PMBOK), but takes this a step further and focuses the project team specifically upon the goals and objectives necessary for the successful completion of high end telecommunications projects. The assigned project manager tracks and coordinates resources, understands and assists the client in identifying project priorities communicating them to the team with controls, checklists, meeting agendas, status reports and other collaboration tools designed to sort out the myriad of conflicts that arise in a complex technical implementation. Using PC3M, the Project Manager communicates and manages priorities building consensus and focus with all elements of the project team, both internal and external.

Our management and technical work processes were developed to confront risks that might impede our ability to complete high quality work on time and within budget. We overcome risk-related obstacles through the application of our risk management processes. This is accomplished with a qualitative and quantitative risk analysis to identify appropriate indicators and measured responses to mitigate any potential issues. Additionally, we utilize a series of project-oriented Gantt Chart tools, Critical Path calculations in our scheduling process to quickly identify resource conflicts, interrelated delays and the impact of changes concerning key components and personnel. Microsoft Project is our tool of choice for our high level tracking tool. In-Progress Reviews (IPRs) will include an evaluation of technical progress, progress on milestones, draft deliverables, and resolution of any issues that arise between IPRs.

The Project Manager will ensure the project objectives are met in due time and budget and in line with the expected quality and result.

EXHIBIT 7

Major Responsibilities of Project Management:

- Definition of and consensus on detailed project scope
- Define solution and all components for the System
- Project change management including all related third party interactions, i.e. software and hardware component suppliers, inside wiring contractors, etc.
- Definition, communication, and management of deployment project plan(s)
- Project coordination and synchronization of implementation activities
- Project reporting and communications
- Project risk management and risk mitigation
- Master Project plan/timeline in Microsoft Project format
- Third-Party Management which include Meeting Planning, Escalation Process Compliance and Supplier Ownership of all third parties.

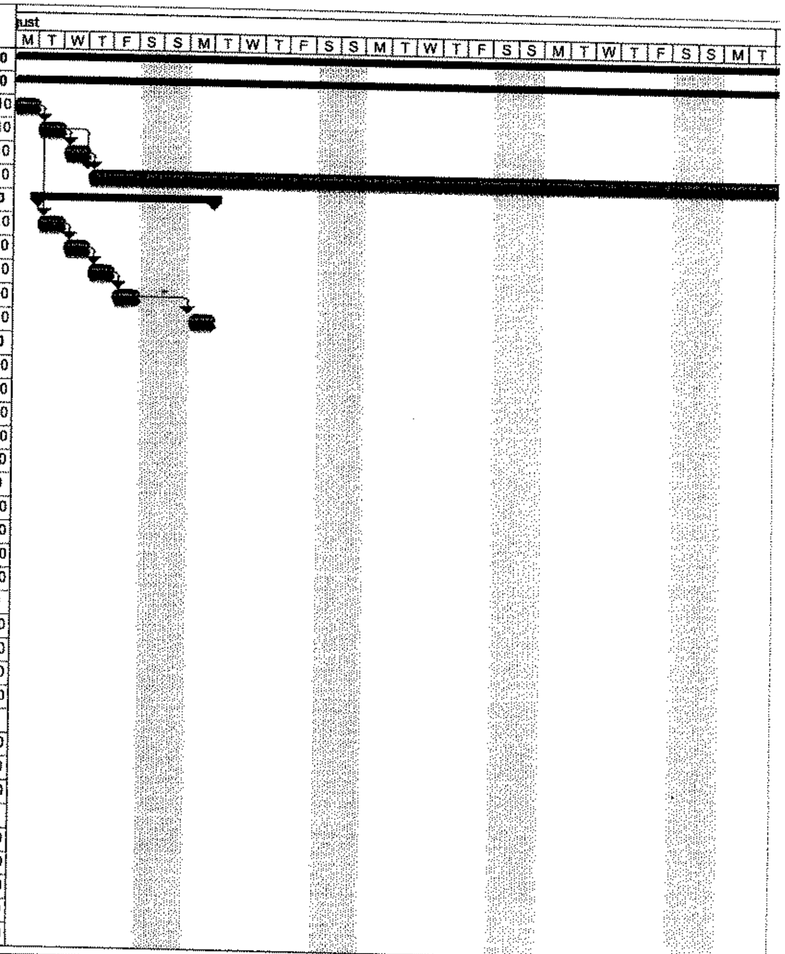
Quality Control

Quality control is seamlessly incorporated into all of our planning and engineering processes to ensure the consistent level of professional services we offer all of our customers. Upon award of a contract, the project manager identifies key areas where quality can be measured and then defines key metrics to measure and control our quality processes. These quality metrics are communicated in our In-Progress Reviews (IPRs), and enable the project manager to quickly identify and mitigate any quality issues before they compound and adversely impact the project deliverables.

Project Plan

The following is a Gantt chart illustrating the activities involved in the installation, configuration and training of the new GCC VoIP Telephony System. We have identified a hypothetical completion date by assuming an award date of August 2, 2010:

ID		Task Name	Duration	Start	Finish
1		**Contract Issued & Notice To Proceed**	**49 days?**	**Mon 8/2/10**	**Thu 10/7/10**
2		**Requirements Analysis**	**33 days?**	**Mon 8/2/10**	**Wed 9/15/10**
3		Kickoff Meeting	1 day?	Mon 8/2/10	Mon 8/2/10
4		Verify Equipment Order	1 day?	Tue 8/3/10	Tue 8/3/10
5		Verify Telco Order Placed	1 day?	Wed 8/4/10	Wed 8/4/10
6		Receive & Inventory Cisco Equipment	30 days?	Thu 8/5/10	Wed 9/15/10
7		**Design**	**5 days?**	**Tue 8/3/10**	**Mon 8/9/10**
8		Review And Design IP Addressing	1 day?	Tue 8/3/10	Tue 8/3/10
9		Review and Design VLAN and Routing Plans	1 day?	Wed 8/4/10	Wed 8/4/10
10		Submit Ip Addressing Changes To Customer For Approval	1 day?	Thu 8/5/10	Thu 8/5/10
11		Review Call Handling Requirements	1 day?	Fri 8/6/10	Fri 8/6/10
12		Develop, Document and Verify Dial Plan	1 day?	Mon 8/9/10	Mon 8/9/10
13		**Program/Build**	**5 days?**	**Thu 9/16/10**	**Wed 9/22/10**
14		Burn-In Cisco Equipment	1 day?	Thu 9/16/10	Thu 9/16/10
15		Install Gateway	1 day?	Fri 9/17/10	Fri 9/17/10
16		General Site Preparation - Verify Environmentals	1 day?	Mon 9/20/10	Mon 9/20/10
17		Verify Data Infrastructure Op - Troubleshoot	1 day?	Tue 9/21/10	Tue 9/21/10
18		Setup & Test Call Mgr And Unity	1 day?	Wed 9/22/10	Wed 9/22/10
19		**Call Manager Configuration & Testing**	**4 days?**	**Thu 9/23/10**	**Tue 9/28/10**
20		Build Call Manager & Unity Database	1 day?	Thu 9/23/10	Thu 9/23/10
21		Program Ext and Feature Sets	1 day?	Fri 9/24/10	Fri 9/24/10
22		Program Voice Applications	1 day?	Mon 9/27/10	Mon 9/27/10
23		Program Unity Voicemail	1 day?	Tue 9/28/10	Tue 9/28/10
24		**Installation And Testing**	**3 days?**	**Thu 9/23/10**	**Mon 9/27/10**
25		Install Call Manager & Unity	1 day?	Thu 9/23/10	Thu 9/23/10
26		Telco Due Date For All New Trunks	0 days	Thu 9/23/10	Thu 9/23/10
27		Test Call Manager	1 day?	Fri 9/24/10	Fri 9/24/10
28		Test PRI/ISDN circuits	1 day?	Mon 9/27/10	Mon 9/27/10
29		**Training**	**3 days?**	**Tue 9/28/10**	**Thu 9/30/10**
30		Prepare/Print GCC Custom Training Doc's	1 day	Tue 9/28/10	Tue 9/28/10
31		Initial End User Training	1 day?	Wed 9/29/10	Wed 9/29/10
32		Set & Detail Phones	1 day?	Thu 9/30/10	Thu 9/30/10
33		**GCC Cutover**	**3 days?**	**Fri 10/1/10**	**Tue 10/5/10**
34		Support And Maintenance	1 day?	Fri 10/1/10	Fri 10/1/10
35		Postcut Onsite Support Day1	1 day?	Mon 10/4/10	Mon 10/4/10
36		System Admin Training Review On Cm, Unity	1 day?	Tue 10/5/10	Tue 10/5/10
37		Final Review of Acceptance Testing Reports	1 day?	Wed 10/6/10	Wed 10/6/10
38		Final Implementation Invoicing	1 day?	Thu 10/7/10	Thu 10/7/10



INSTALLATION

Thé objective is to provide a successful installation of the proposed solution. GTA is responsible for the installation and commissioning of:

- Unified Call Manager
- Server
- Media Gateway
- Instruments
- Power over Ethernet (POE) Switches

Based on the assumptions made regarding solution components, GTA has developed an estimated effort and determined the resources necessary to implement and commission the various components into GCC's network. The GTA VoIP Team proposes to perform a comprehensive site survey for the purpose of establishing a detailed understanding of the installation environment at the various sites where equipment is to be installed.

After each site is surveyed, the GTA VoIP Team will provide a site plan that describes the site specific installation plan that includes location of equipment to be installed. Drawings and digital photos, when appropriate, will be used to provide installation plan details and to memorialize the pre and post installation environment. Additionally, the installation plan will address specific plans for assuring that installation is accomplish in a manner that is astatically pleasing. The installation plan will address any site preparation that will be required as a result of equipment installation. The ultimate goal of the installation plan is to describe the installation process that will be used to install the VoIP equipment in an astatically pleasing fashion and any activity required to restore the site after installation is has been completed.

The plan will be provided to GCC's designated technical authority for approval prior to beginning installation and construction.

The installation activities to support VoIP will include installation of trunk lines, switching and server equipment, and a variety of terminal and premises equipment. The intent of the installation plan is to perform all of these tasks in a parallel to reduce the overall prográm installation interval. When the trunk lines are ready; the switches and servers will be ready; the terminal equipment will be ready and we will commence configuration and integration, test and turn-up of the system. We will focus on the bringing users on line in as the infrastructure becomes available. We will concentrate our work to provide service on a building by building or area by area basis.

Integration, End-to-End Test Activities & Solution validation

GTA TeleGuam will perform ongoing integration, testing and solution validation of the solution components.

- System Delivery
- Installation of Systems - software and associated hardware
- Configure and test System, ready for customer acceptance

User Acceptance Test (UAT)

The goal of UAT, is to test if the solutions deployed is working in GCC's environment according to GCC requirements and to achieve acceptances. GTA will support GCC during the UAT testing.

Training

GTA TeleGuam will provide the following training services associated with the system. The intent is to provide knowledge transfer to GCC staff on the system that has been installed. The services and activities:

- Formal classroom training on introduction and operation of individual components
- On-the-job training, with Integrator experts providing knowledge transfer to GCC staff on the operation of the solution, first customer contact maintenance, and administering of the systems.

Launch Support & Operational Handover

In order to guarantee a successful start of the new service, GTA TeleGuam will support GCC, ensuring a smooth handover of responsibilities and knowledge at the end of the project. Such assistance will make sure that experienced personnel perform all daily tasks. During the handover GTA TeleGuam shall provide relevant documentation including handover document and will conduct a handover session in which all still open issues will be discussed

Technical Support and Maintenance Services

The support services shall include:

a) Tier 1 support – Will be provided by GCC IT staff.
b) Remote Tier 2 helpdesk for technical support service by GTA TeleGuam
c) Minor software updates/ patches with bug fixes.
d) Technical analysis and resolution of problems, escalated by lower support tier.
e) Technical consultation for questions and queries related to the supported system.
f) Interface with the suppliers of third party embedded in the system. (when applicable)

EXHIBIT 8

Remote Support

GTA's VoIP Vendor supports many customers remotely through VPN access to the customer's network. GTA's VoIP Vendor considers customer network information as classified. GTA's VoIP Vendor encrypts this information and only assigned engineer and the engineering director have access to the network information.

Local Support

GTA's VoIP Vendor has contracted with a Cisco Certified Internetworking Engineer ("CCIE") on Guam to respond to any major issues that cannot be addressed by GTA engineers or remote VPN access.

Trouble Response Approach

Our response package for first year warranty and maintenance service is explained below. All equipment is covered by manufacturer warranty. Cisco Smartnet is included for the materials listed in the bid response. Critical equipment is covered by Cisco's next day SmartNet coverage.

GTA's VoIP Vendor's normal approach for warranty service is to offer a pre-paid block of hours for all services. With a pre-paid block of hours an agreed upon Service Level Agreement (SLA) is established for response; in this case reflecting the 1-hour onsite requirement for critical issues, 24 hours per day, seven days a week. GTA's VoIP Vendor will guarantee remote troubleshooting and diagnosis within 30 minutes of any tier 2/3 trouble call. Technical assistance in excess of the Block of Hours are paid for at the prevailing technical or consulting rate. Hours can be used for any function desired by GCC.

The use of a Block of Hours in place of a standard Service Contract has many advantages: GCC does not pay for services it does not receive. As the end of the year approaches any remaining hours can be used for training, network assessment, consulting, small projects, etc.

The overall cost is lower to GCC than having to pre-pay for anticipated services on many different applications. The cost of service contracts for the VoIP system, VoiceMail, VTC and Network problems are cumulatively more expensive than having a single block of hours to cover them all. Disputes regarding whether an event is covered by warranty or is a chargeable action are mitigated.

GTA's VoIP Vendor is offering a 50-hour Block of Hours for GCC. This can be adjusted as needed prior to contract execution. A standard Service Agreement is also available if desired. The standard procedure for response to a catastrophic network or voice system failure or critical component failure is to immediately dispatch a field technician to the site while a second engineer logs in remotely to assess and troubleshoot the outage. GTA's VoIP Vendor management will be notified once the engineers are enroute. GTA's VoIP Vendor will begin notification procedures to the agreed upon key contacts with GTA and GCC management. Hourly updates regarding the status of restoration efforts will be provided as the troubleshooting process is executed. GTA's VoIP Vendor will implement a GCC approved escalation procedure that includes the Cisco TAC and more senior management if specific time limits for the outage are surpassed.

Commercial Terms

This quote is subject to GTA TeleGuam's Master Services Agreement of sale ("MSA"). The ordering of products or services by the purchaser ("Customer") (including through placement of any written purchase order, work order or similar document) shall constitute acceptance of the MSA. GTA TeleGuam shall not be deemed to have accepted (including by implication or conduct) any other terms or conditions unless GTA TeleGuam has explicitly agreed in writing to be bound to such terms and conditions. The MSA shall prevail over any terms and conditions stated or referenced in any purchase order work order or other similar document issued by the Customer and those stated or referenced terms and conditions shall be of no force or effect whatsoever.

EXHIBIT 9

PLATINUM LEVEL

SERVICE LEVEL AGREEMENT

Between
GTA TeleGuam LLC
and
Guam Community College

Commencing ________ 2010
Expiring __________ 2011
with option to renew

Confidential and Proprietary.

Submitted to:
Guam Community College

Submitted by: Jennifer Sgambelluri
Approved by: Roland Certeza
Authorized Signature: John J. Kim

TeleGuam Holdings, LLC LLC
624 N. Marine Corps Drive
Tamuning, GU 96913
Tel: 671.644.0116
Mobile: 671.488.5522
Fax: 671.644.0103

EXHIBIT 10

Purpose

The purpose of this Platinum Level Service Level Agreement (SLA) is to formalize an arrangement between GTA TeleGuam, LLC and Guam Community College to define the services to be rendered, describe the kind of support to be provided, outline roles and responsibilities of all parties, and articulate the consequences and boundaries of unacceptable service performance. This SLA can evolve other time, with additional knowledge of the client requirements, as well as the introduction of new services into the support portfolio provided to Guam Community College.

Descriptions of Service

GTA TeleGuam's solution includes the following specifications as part of the monthly recurring cost:

- GTA TeleGuam operates a 24 x 7 x 365 technical support hotline.
- GTA TeleGuam targets 99.999% network uptime on the local access.
- Landline Services
- VoIP PBX Maintenance
- Managed Router Service
- **Anything else?**

Description of Support

This is support provided by the GTA TeleGuam Call Center when it receives a call from Guam Community College. Our 24 hour, 7 days a week, 365 days a year Customer Service Department will handle calls professionally, accurately, and will provide timely updates on your issues.

GTA TELEGUAM CALL CENTER	
AVAILABILITY	**PHONE NUMBER**
24 x 7 x 365	(671) 644-4482

When reporting service problems, please provide the following information so that GTA TeleGuam may service you more efficiently:

- Telephone/Circuit Number
- Point of Contact Information
- Description of Problem

If this level of support cannot resolve the problem, the trouble ticket is passed to the Level 2 support, which is our Technical Assistant Center.

Service Standards

The following table explains GTA TeleGuam response and remedy times once a call has been placed to the GTA TeleGuam Call Center and a trouble ticket has been created:

PLATINUM SERVICE LEVEL AGREEMENT		
EVENT	**COMMON ACCESS***	**NON-COMMON ACCESS****
1. Contact Customer	0-2 Hours	0-2 Hours
2. Send Technician to:		
Central/Remote Office	0-2 Hours	0-2 Hours
Customer Premise	0-2 Hours	Subject to access.
3rd Party Facility Access	Not Applicable	Subject to vendor availability
3. To Remedy Issue		
Not related to cable damages	2-4 Hours	Subject to vendor availability
Cable damage by 3rd Party	Based on severity	Subject to vendor availability

***Common Access** is defined as GTA having unfettered access with no 3rd party coordination required.

****Non-Common Access** is defined as GTA having to coordinate with Military base personnel, building security officers etc. In these instances, after hours and overtime charges may apply by 3rd Party. GTA overtime is included as part of this SLA. (Military Bases, APAC, ITC Bldg, etc.)

Technician Availability
24 hours, 7 Days a Week, Holidays Included

Overtime Charges
Inclusive of SLA

Escalation Procedures

If no response within 30 minutes of reporting the trouble to Call Center, please follow escalation procedure.

Contact Person	Title	Work	Mobile	Email
1. Business Sales		644-1000		businesssales@gta.net
2. Jennifer Sgambelluri	Sr. Account Manager	644-0116	488-5522	jsgamby@gta.net
3. Roland Certeza	EVP, Sales & Mktg	644-0129	488-5596	rcerteza@gta.net
4. Colin Kerber	VP, IP Broadband	644-4351	488-0049	ckerber@gta.net
5. Carl Leon Guerrero	EVP, Operations	644-2000	483-2275	carl@gta.net

Overtime

GTA will not charge for any GTA technician overtime. Any charges billed by Third Party vendor, will be assessed to Customer. Prior approval from Customer will be required before scheduling Third Party after hour access.

Confidential and Proprietary.

PLATINUM LEVEL

SERVICE LEVEL AGREEMENT

Between
GTA TeleGuam LLC
and
Guam Community College

Commencing _________ 2010
Expiring ___________ 2011
with option to renew

Submitted to:
Guam Community College

Submitted by: Jennifer Sgambelluri
Approved by: Roland Certeza
Authorized Signature: John J. Kim
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Mobile: 671.488.5522
Fax: 671.644.0103

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Technician Availability		
24 hours, 7 Days a Week, Holidays Included		
Overtime Charges		
Inclusive of SLA		

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Overtime

GTA will not charge for any GTA technician overtime. Any charges billed by Third Party vendor, will be assessed to Customer. Prior approval from Customer will be required before scheduling Third Party after hour access.

Credit Allowance for Service Interruptions

An interruption period starts when an inoperative service is reported to the GTA TeleGuam Call Center, and ends when the service becomes operative. A credit allowance is not given when interruption is due to fault at the customer's end. The Customer will receive a credit for interruptions of 2 hours or more. The rate is 1/1440 of the fixed monthly service charges for each period of 30 minutes or major fraction thereof. Service credit is based on monthly charge of interrupted service. Credit allowance shall not exceed the monthly service charges. The allowance will be payable in the form of a credit to the Customer's account. Credits shall not apply to equipment charges.

Exclusions

GTA TeleGuam cannot be held responsible for, and Credit Allowance will not be issued, when the service interruption is the result in whole or in part of one of the following causes:

- Any act or omission on the part of the Customer, any third party contractor or vendor, or any other entity over which the Customer exercises control or has the right to exercise control.
- The Customer's applications, equipment, or facilities.
- GTA TeleGuam's or the Customer's scheduled maintenance.
- Any event or occurrence that results in "No Trouble Found" resolution to trouble tickets.
- Interruptions where the Customer elects not to release the service for testing and repair, and continues to use it on an impaired basis.
- Interruptions during any period when GTA TeleGuam is not allowed access to the Customer premises where affected access lines are terminated.
- Events beyond the reasonable control of GTA TeleGuam including, but not limited to: natural disasters, cable cuts, government acts and regulations, and national emergencies.

Duration

This Service Level Agreement may only be applicable to contracted services. Commencement of SLA requires both signatures from Guam Community College and GTA TeleGuam, LLC. This Service Level Agreement will expire at the same time as contract between Guam Community College and GTA TeleGuam LLC expires or is voided.

Existing Circuits & Services

GCC Circuits	Circuit Type	Point A	Point B

Key Personnel

Name	Work	Email

Agreed and Acknowledged by Authorized Representative:

GTA TeleGuam | **Guam Community College**

Date | Date