IPCOMM2006
September 25-27 • Gaylord Opryland • Nashville, TN

IPT CASE STUDY

IPT from Pilot to 5,000 Stations

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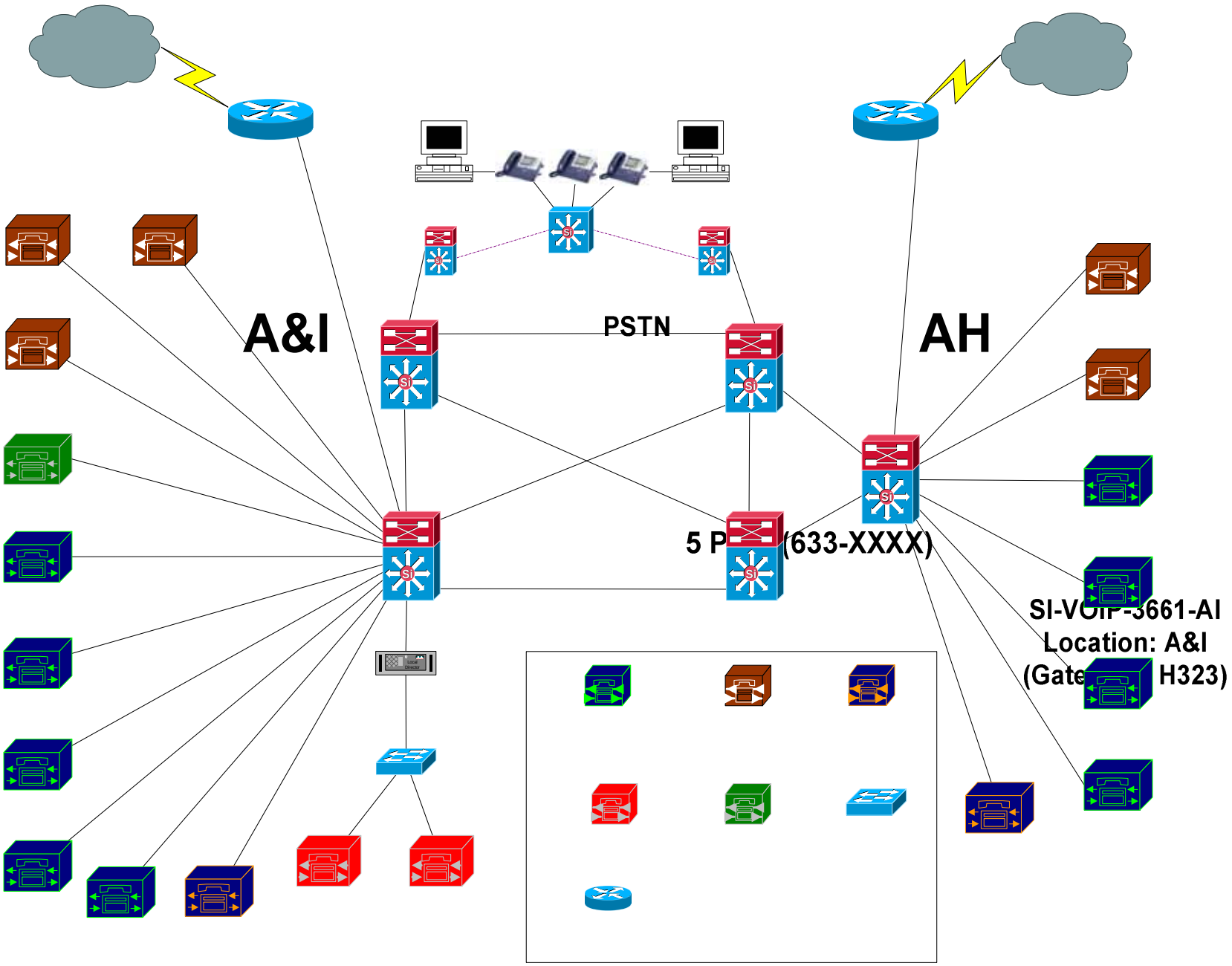
What Will Be Covered

- Business Case for VoIP at the Smithsonian
- Deployment Objectives
- Deployment Process
  - Network Upgrade
  - VoIP Deployment
- Operations and Maintenance
- Key Points to Take Home


A&I
PSTN
AH
5 P (633-XXXX)
SI-VOIP-3661-AI
Location: A&I
(Gate H323)
SI-VMBU01
Location: A&I
(Unity Voice Mail)
SI-VMEXCH01
Location: A&I
(Unity Voice Mail)

Business Case

Smithsonian Starting Point-2001

- Heterogeneous & proprietary mixture of service, equipment, telephone instruments, systems
- Telecommunications “islands” with little or no interconnectivity
- Arbitrary numbering & dialing patterns
- Inconsistent features, functions, & applications
- High level of old/obsolete systems
- Inconsistent levels of service & support
- Inability to support SI business functions
- Old physical infrastructure, e.g., Cat 3 or unrated
- Homogeneous data environment (Cisco) modernized physical plant, e.g., Cat 5/5E
- Components of Data Network *End of Life 2002-2005*

Modernization Goals

- Cost
  - Reduce operational & maintenance cost
  - Improve cost accountability & tracking
- Service
  - Consolidate all systems into a single entity
  - Establish system SLAs within SI community
  - Improve MAC process
  - Improve response & resolution intervals
- Technical
  - Improve network redundancy & failover
  - No single points of failure (discussion later)
  - Centralization of applications & management
  - Service offerings to any SI location based on business requirements
  - Central control with limited localized administration
  - E911 information provision
  - End user-controlled moves

Architecture Evaluation Process*2001

1 = Best 2 = Better 3 = Good	CO-Based Solution	PBX-Based Solution	IP-Based Solution
Standards-Based	2	3	1
Scaleable	1	2	1
Migration from Current to New Solution	2	3	3
Features, Functions, & Applications	3	1	2
Management & administrative control	3	2	1
Reliability & availability	1	2	2
Mature, proven technology	1	1	2

Requirements

Project Cost Elements	No Action	Average PBX-Based Solution	CO-Based Solution (Centrex®)	IP-Based Solution
Capital Cost				Includes Data Switches
Design and Planning	$ -			
Common Equipment	$ -			
Telephone Sets	$ -			
Voice Messaging	$ -			
Call Center Functionality	$ -			
Maintenance	$ -			
Installation/Integration	$ -			
Subtotal	$ -	**$ 8,527,478**	**$ 7,050,000**	**$ 11,285,000**
Indirect Project Cost				
Cabling Estimates	$ -			
Installation of PSTN	$ -			
PSTN Recurring Cost	$ -			
SI Staff	$ -			
Dual Operations Service	$ -			
Subtotal	$ -	**$ 6,954,000**	**$ 7,455,000**	**$ 4,381,500**
Total Project Cost	**$ -**	**$ 15,481,478**	**$ 14,505,000**	**$ 15,666,500**

Life Cycle Cost

	No Action	Average PBX-Based Solution	CO-Based Solution (Centrex®)	IP-Based Solution
Year 1	$ 4,208,000	$ 7,640,739	$ 6,707,500	$ 6,541,250
Year 2	$ 4,208,000	$ 6,112,591	$ 4,832,000	$ 5,233,000
Year 3	$ 4,208,000	$ 5,468,668	$ 6,173,500	$ 4,083,250
Year 4	$ 4,208,000	$ 3,940,250	$ 4,832,000	$ 2,775,000
Year 5	$ 4,208,000	$ 3,940,250	$ 4,832,000	$ 2,775,000
Subtotal	**$ 21,040,000**	**$ 27,102,498**	**$ 27,377,000**	**$ 21,407,500**
Year 6	$ 4,208,000	$ 5,940,250	$ 4,832,000	$ 2,775,000
Year 7	$ 4,208,000	$ 3,940,250	$ 4,832,000	$ 2,775,000
Year 8	$ 4,208,000	$ 3,940,250	$ 4,832,000	$ 2,775,000
Year 9	$ 4,208,000	$ 3,940,250	$ 4,832,000	$ 2,775,000
Year 10	$ 4,208,000	$ 3,940,250	$ 4,832,000	$ 2,775,000
Total	**$ 42,080,000**	**$ 48,803,748**	**$ 51,537,000**	**$ 35,282,500**

Deployment

Deployment Objectives

- Enhance network reliability to support VoIP operations
- Implement a “cookie-cutter” approach to minimize deployment time, miscommunication, and implementation errors
- Implement process that can quickly and accurately transfer information from customers to VoIP engineers
- Stay on schedule to prevent cost overruns

Initial Activities

- Developed design basis for network improvements and VoIP operations
- Developed deployment processes and procedures
- Tested technical design and operation in a technology pilot with IT users
- Tested deployment processes and procedures in a pilot using non-IT users

VoIP Deployment Process

- The SI VoIP deployment process consisted of 2 phases: data deployment ("forklift" upgrade) and voice deployment

- Data Deployment
  - Survey of LAN/WAN network equipment to determine equipment upgrades required to support the VoIP phone system
  - Scheduled cutover to upgraded network at least one month prior to beginning phone deployment

- Voice Deployment
  - Obtain user configuration information for the design of phone systems for each organizational office within the building.
  - Deploy VoIP phone system and disconnect old phone system

Data Deployment

Step 1: Site survey to document equipment and infrastructure (power and fiber/cable) changes needed to meet design requirements

- Site Survey recommendations reviewed by all stakeholders for approval
- Site survey became implementation roadmap for the network upgrade

Step 2: Equipment configured, installed and tested at least one month prior to phone installation

- Configured with separate voice and data VLANs to enhance security posture
- Identified existing equipment that required switch ports set 10/half (all other ports set 100/full)

Step 3: Final As-Built document for each location documents all aspects of the upgraded network

- electrical, cable/fiber
- switch and router configuration
- equipment locations and serial numbers
- test and acceptance documentation

Voice Deployment

Step 1: Kick-off meeting with office POCs

- Review deployment policies, information gathering tools, and schedule

Step 2: Completion of the User Data Collection Spreadsheet

- User Spreadsheet completed electronically by customer POC
- Provides an inventory of phones required, phone locations, user names
- User Spreadsheet used to confirm requisite cable jacks are present for all requested phone locations

Step 3: Completion of the Office Questionnaire

- Gathers information from customer on call forwarding of old numbers to new VoIP numbers, new phone configuration, and voicemail

Voice Deployment cont…

Step 4: Call flow developed and submitted to customer for approval

- Call flow is a diagram is of the information provided by the customer
- Phone configurations begin following approval of call flow

Step 5: Phone configuration

- MAC addresses are scanned into the spreadsheet for each phone
- Spreadsheet imported into the Installation Database
- Database automatically assigns phone numbers
- Information from database is then bulk imported into CallManager (the brains of the VoIP network) and the voicemail system

Step 6: Phone installation

- Distribute new numbers to customers and for employee directory update
- Execute forwarding and disconnect orders for legacy phone numbers
- Provide customer training on new phones
- Install phones and provide post-installation support

Customer Training

Training on the new phone system was delivered to customers in three ways:

- Web-based training materials
  - Project website contained link to VoIP phone function tutorial, VoIP Users Reference Guide, and training slides used in hands-on training
- Hands-on VoIP phone training
  - Trainer provided users with a hands-on overview on the operation of the VoIP phone
  - Training provided typically provided within 5 days of installation
- Information provided at time of installation
  - Mouse-pad containing step-by-step directions for commonly used phone features such as ad-hoc conferencing and call forwarding
  - Information sheet with directions on how to setup voicemail accounts and web access accounts (CCM User)

Operations and Maintenance

Telecommunications Staffing

- 5 FTEs for VoIP support
  - 1 Senior VoIP engineer
  - 1 Mid-Level VoIP engineer
  - 2 Junior VoIP engineers perform new installations and resolve service tickets (~180 VoIP-related tickets per month)
  - 1 CCIE with converged network expertise
- 3 FTEs for legacy support
  - Billing, provisioning, miscellaneous non-VoIP services
- Lines of communication and responsibilities must be clearly defined for network and telecommunications staff

Operating a Converged Network

Network Monitoring

- Voice packets much more susceptible to packet loss so WAN/LAN links must be monitored for saturation

Network Change Control

- Formal process required due to potential for impact on VoIP
  - Changes to switches can invalidate ER911 information and must be controlled

Lab Testing

- All software upgrades and patches must be tested & verified prior to introducing into the production environment
- Security and firewall rules MUST be tested with VOIP

Security

- Separate VLANs required for voice and data
- Disable unnecessary services on servers
- Perform a penetration test to identify and minimize network vulnerabilities
- Ensure password protection is strictly enforced

Key Points to Take Home

- Processes, procedures, roles and responsibilities should be battle-tested in a pilot test before using in full-scale deployment
- Think through the design basis for the network and document everything
  - E.g., telecommunication needs during extended power outages or loss of network connectivity
- Make use of automated tools such as bulk import tools to maximize efficiencies and minimize data entry errors
- Communicate with the customers and make them stakeholders in the deployment process
- Strongly discourage customer changes after the configuration process has begun and do not allow changes once the deployment is underway
  - Minimize the time period from receipt of customer information until phones are installed
- Ensure that a VoIP lab capable of replicating the production environment is in place

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QUESTIONS?

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Smithsonian Institution

Appendix

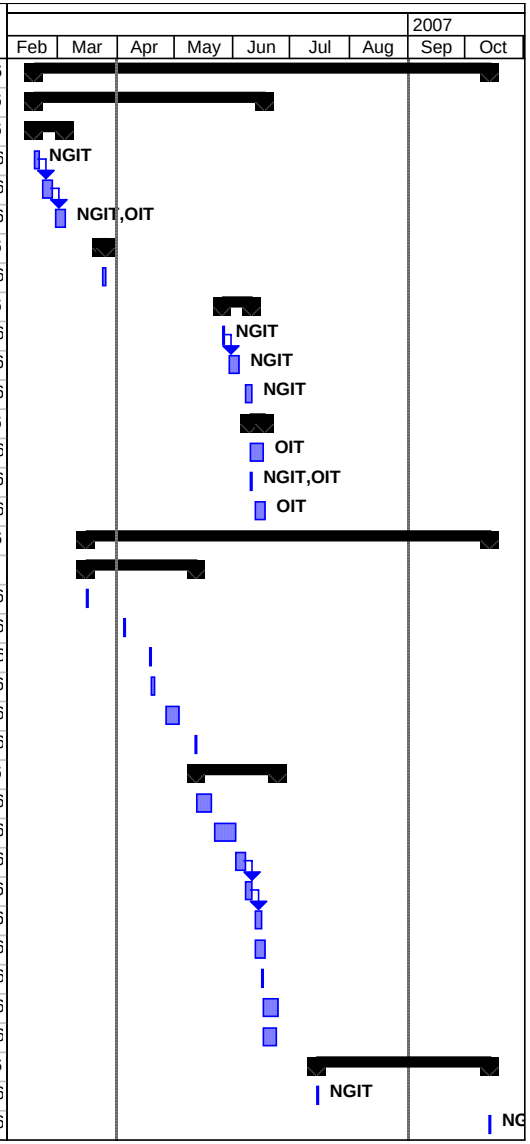
SI Network Design Basis for VoIP

- Network design changes were based on enhancing redundancy/reliability and minimizing single points of failure
  - Redundant power supplies used in all switches (have never had a power supply failure)
  - Access switches dual-homed to distribution switches
  - Redundant distribution switches used in all buildings
  - Redundant core switches (4)
  - UPS with minimum up-time of 30 minutes provided for all switches
  - When feasible, emergency diesel power used for power circuits
  - Separate electrical circuits used for UPS and switch power

SI VoIP Network Design Basis

- VoIP network design based on enhancing redundancy/reliability and minimizing single points of failure
  - Three levels of redundancy provided to CallManagers
  - Redundant servers used for voicemail and all other VoIP applications
  - SRST routers used for remote locations to provide basic telcom service in the event network connectivity is lost
  - Redundant VoIP servers for the Washington DC cluster are located in two separate locations on the Mall to provide an additional layer of protection

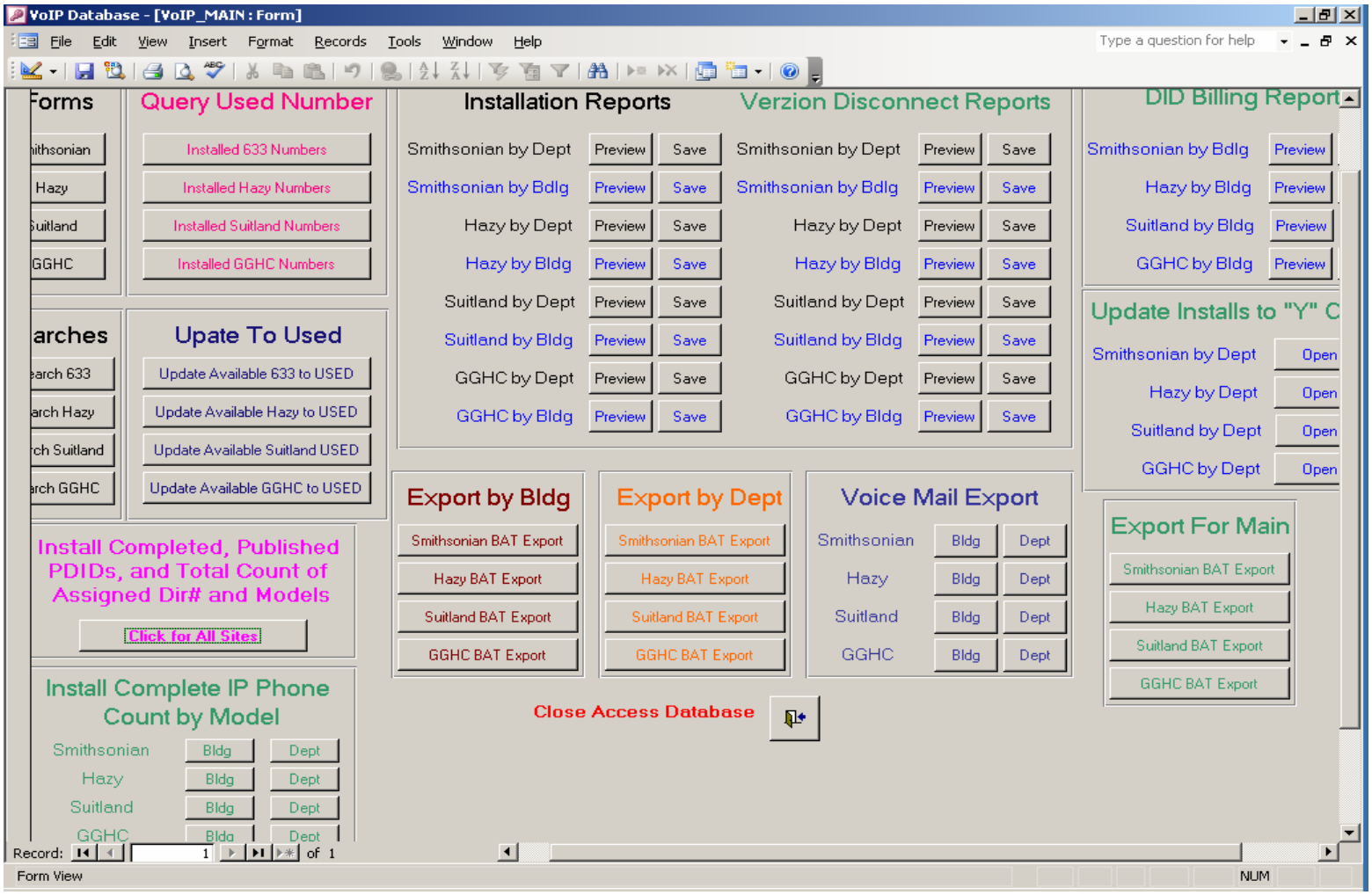
ID	Task Name	Duration	Start	Finish
1114	**Capital Gallery Phase 1 (200 phones - floors 3 and 4)**	**172 days**	**Thu 2/16/06**	**Fri 10/13/06**
1115	**Data Deployment**	**87 days**	**Thu 2/16/06**	**Fri 6/16/06**
1116	**Data Infrastructure Plan & Preparation**	**12 days**	**Thu 2/16/06**	**Fri 3/3/06**
1117	Perform site survey	2 days	Thu 2/16/06	Fri 2/17/06
1118	Document survey findings	5 days	Mon 2/20/06	Fri 2/24/06
1119	Review & approve recommendations	5 days	Mon 2/27/06	Fri 3/3/06
1120	**Bill of Materials (BOM) Development**	**1 day**	**Fri 3/24/06**	**Fri 3/24/06**
1121	Order Network Equipment	1 day	Fri 3/24/06	Fri 3/24/06
1122	**Data Infrastructure Deployment**	**11 days**	**Fri 5/26/06**	**Fri 6/9/06**
1123	Receive equipment at NGIT staging area	1 day	Fri 5/26/06	Fri 5/26/06
1124	Configure equipment	5 days	Mon 5/29/06	Fri 6/2/06
1125	IDF Installation	3 days	Wed 6/7/06	Fri 6/9/06
1126	**Post Data Migration**	**6 days**	**Fri 6/9/06**	**Fri 6/16/06**
1127	Post-implementation monitoring	5 days	Fri 6/9/06	Thu 6/15/06
1128	NOC turnover	1 day	Fri 6/9/06	Fri 6/9/06
1129	Document final design for as-built	5 days	Mon 6/12/06	Fri 6/16/06
1130	**Voice Deployment**	**153 days**	**Wed 3/15/06**	**Fri 10/13/06**
1131	**Voice Configuration Planning**	**42 days**	**Wed 3/15/06**	**Thu 5/11/06**
1132	Identify Museum and Department POCs	1 day	Wed 3/15/06	Wed 3/15/06
1133	Deployment Presentation/Distribute Phone Data Sheet	1 day	Tue 4/4/06	Tue 4/4/06
1134	Return Phone Data Sheet	1 day	Mon 4/17/06	Mon 4/17/06
1135	Workshop Briefing	2 days	Tue 4/18/06	Wed 4/19/06
1136	Configuration Workshop	5 days	Wed 4/26/06	Tue 5/2/06
1137	Return Office Questionnaire	1 day	Thu 5/11/06	Thu 5/11/06
1138	**Voice Deployment**	**31 days**	**Fri 5/12/06**	**Fri 6/23/06**
1139	Prepare Call Flow Diagrams	6 days	Fri 5/12/06	Fri 5/19/06
1140	Division Sign Off (Call Flow)	9 days	Mon 5/22/06	Thu 6/1/06
1141	Update Installation Database	3 days	Fri 6/2/06	Tue 6/6/06
1142	Update CallManager	3 days	Wed 6/7/06	Fri 6/9/06
1143	Setup Voicemail Acct	3 days	Mon 6/12/06	Wed 6/14/06
1144	Training	5 days	Mon 6/12/06	Fri 6/16/06
1145	Pre Installation Verification	1 day	Thu 6/15/06	Thu 6/15/06
1146	Installation of Phones	6 days	Fri 6/16/06	Fri 6/23/06
1147	Call Forward	5 days	Fri 6/16/06	Thu 6/22/06
1148	**Voice Post-Deployment**	**66 days**	**Fri 7/14/06**	**Fri 10/13/06**
1149	Remove call forwarding & change to intercept message (after 10 da	1 day	Fri 7/14/06	Fri 7/14/06
1150	Remove intercept (after 90 days)	1 day	Fri 10/13/06	Fri 10/13/06

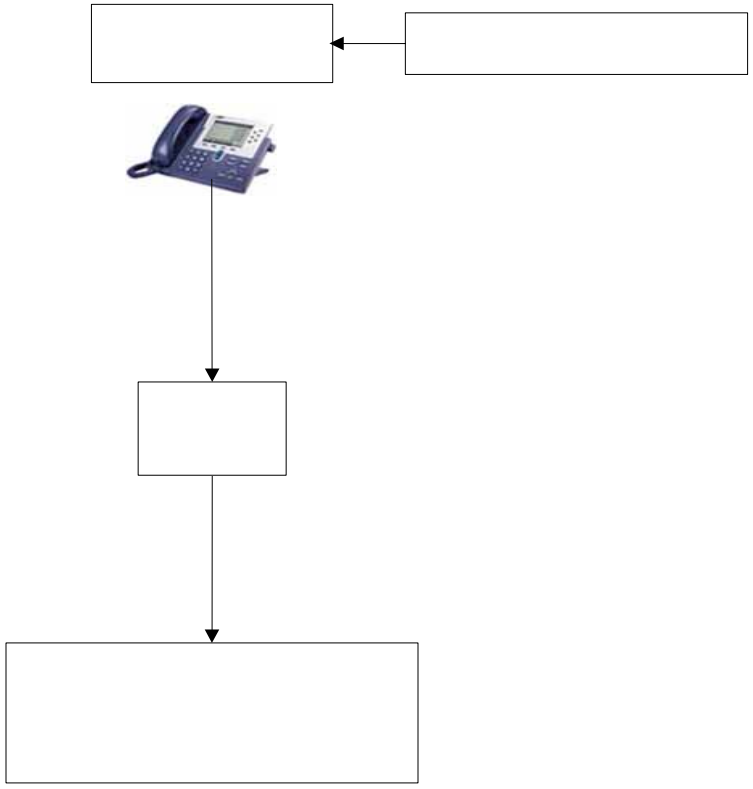


User Spreadsheet

Last Name	First Name	Ph. #	Network ID	Bldg	Office	Room	VM	Class of Service	Org Code	WM	Non-User Description	PC
Barber	Carolyn	357-2944	barberc	Quad	Contr. Mem	3156	Y	4	182000	N		Y
Barton	Jennifer	357-2945	bartonj	Quad	Contr. Mem	3150	Y	4	182000	N		Y
Brouse-Long	Laura	357-3075	brouselongl	Quad	Contr. Mem	3153	Y	4	182000	N		Y
Butler	Marguerite	357-2175	butlerm	Quad	Contr. Mem	3150	Y	4	182000	N		Y
Kelly	Joyce	633-9185	kellyj	Quad	Contr. Mem	3156	Y	4	182000	N		Y
Stewart	Katharine	357-2636	stewartk	Quad	Contr. Mem	3146	Y		182000	N		Y
Walker	Mary Anne	786-3162	walkerma	Quad	Contr. Mem	3146	Y	4	182000	N		Y
Vacant Ops Mgr		786-3163		Quad	Contr. Mem	3154	Y	4	182000	N		Y
Vacant JSS Mgr		357-2966		Quad	Contr. Mem	3152	Y	4	182000	N		Y
Conf. Room		633-8938		Quad	Contr. Mem	3157	N	4	182000	N	CM Conf Room	Y

Installation Database





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Mousepad

How to Place a Call on Hold
While on a call:
- Press **Hold** soft key. (Number flashes on LCD screen.)

To resume a call:
- Select number on LCD screen and press **Resume** soft key.

How to Forward All Calls
- Press **CfwdAll** soft key. After 2 beeps, dial the 5-digit extension of the location the calls will be forwarded. Press **CfwdAll** again to cancel.

How to Answer 2nd Call
While on a call and line beeps:
- Press **Answer** soft key to speak with 2nd caller.

To **end** 1st or 2nd call:
- Select line. Press **EndCall** soft key.

To **resume** 1st or 2nd call:
- Select line. Press **Resume** soft key.

How to Transfer
While on a call:
- Press **Trnsfer** soft key. (Do not place a call on hold first.)
- Dial number to which you want to transfer the call.
- Press **Trnsfer** again…OR…
- If the party answers, announce the call and press **Trnsfer**.

How to establish a Conference Call
- While on a call, press the **more** soft key.
- Press **Confrn** soft key. Dial phone number and wait for answer.
- Press **Confrn** soft key to connect calls.

How to use Call History
- Press **directories** button.
- Scroll and highlight Missed, Received, or Placed Calls.
- Press **Select** soft key to view history.
- Press **Clear** soft key to delete all Call History.

How to use Corporate Directory
- Press **directories** button.
- Highlight **Corporate Directory**.
- Press **Select** soft key.
- Enter information using dialing pad and press **Search** soft key.
- Scroll through list to find name and number.

How to Use Dial Plan

Desired number to Dial	Number Dialed from IP Phone
IP Phone User	*XXXXX*
Non-VoIP SI Phone (applies to Wash DC ONLY)	*XXX-XXXX*
Outside Calls (local)	*9-XXX-XXX-XXXX*
Long Distance Calls	*9-1-XXX-XXX-XXXX*

HOW TO CONFIGURE FAST DIALS

1. From browser enter http://ccmuser.si.edu
2. Enter user ID and password
3. Select **Personal Address Book**
4. Select Add New Entry and enter contact information
5. After **Insert** and **Update**, select **Return to Search**
6. Select **Personal Address Book** and then the **Fast Dials** Icon next to name
7. Choose **Index** location and **Directory Number** to fast dial
8. Select **Insert**

VOICEMAIL INTERNAL
- Select Line and Press **messages** button
- Enter password
- See below for Voicemail "during message" commands

VOICEMAIL EXTERNAL
- Dial your VoIP phone number and press *
- Enter your 5 digit phone number (*ID*) and password
- See below for Voicemail "during message" commands

1 – restart	2 – save	3 – delete	4 – slow play	7 – rewind	8 – pause	9 – fast forward	# - skip

To change your Personal Greeting press 4. Then press 1 for Greetings and Transfer.

For assistance, please call OCIO Helpdesk at (202) 633-4000 or open a HEAT service ticket.