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VoIP Service

Voice over IP (VoIP) will be the telephony offering for any new building construction projects. Furthermore, VoIP can be a solution consideration to any service request when it appears that VOIP might be the appropriate alternative. Appropriate alternatives would include but not be limited to remote locations, temporary housing, conference or short term event use, departments willing to pay the implementation fees, planned migration projects or other targeted marketing opportunities. For any questions regarding VoIP service, please contact IS Telecommunications Services at 621-8999.

Standard Features

As default, every prime VoIP line will be programmed with an Additional Call per Line. This provides the ability to have two simultaneous calls – incoming, outgoing or both – with the third incoming call receiving busy treatment (busy signal or voice mail). Since this is a standard feature, there is no monthly-recurring charge as long as it is on the prime line. End Users can elect not to have this feature. Any changes to the number of Additional Calls per Line will require the issuance of a Service Order and charges will apply, unless requested at time of installation.

Each button on a VoIP set requires a unique telephone number. Therefore, all multiple appearances of a telephone number will appear on one physical button regardless of the number of appearances. If a user needs the ability to take multiple simultaneous incoming calls and wants each incoming call to appear on individual buttons, different telephone numbers programmed with Call Forward No Answer/Call Forward Busy will need to be programmed and the Line Text Label can be changed to make it appear that all of the lines are the same number. See Figure 1 for an example and rates.

Call Forward All Calls can be forwarded to any number as allowed by the long distance restriction level. Call Forward All Calls is for the primary line only, and is end-user definable and can be changed at the end user’s discretion. However, Call Forward Busy and Call Forward No Answer cannot be forwarded to an off campus number. Call Forward Busy and Call Forward No Answer are not end-user definable and can only be programmed by the Switch Room. Any other lines can be forwarded (All Calls) by logging into your CCM User account for your phone. Any changes to Call Forward Busy and Call Forward No Answer will require the issuance of a Service Order and charges will apply.
Standard Features (cont.)

Call Forward No Answer ring duration times can vary. The default is 18 seconds. Any changes to the default will require the issuance of a Service Order and charges will apply.

Calling number display is a standard VoIP feature and carries no additional monthly charge.

Caller ID block (Privacy) is available on a per call basis by dialing "*67" before placing the call. Caller ID block (Privacy) is not available on a per line basis.

Long distance restrictions are programmed at the telephone set level, not the telephone line level. On each individual telephone set, a shared telephone line will carry the same long distance restrictions as the primary line of that set.

Line Class Code (LCC) indication is not required in the VoIP environment. In the VoIP environment, all phones have the ability to accept an FRS account to bill long distance charges by entering "*95" before dialing the long distance number. In addition, all VoIP phones have the ability to override the long distance restriction with the use of an Individual Account Code (IAC). For instructions on how to dial long distance calls, please refer to the VoIP phone training presentation (pdf) available at http://uits.arizona.edu/index.php?id=691.

Transferring of calls requires the ability to access dial tone. If an end user has opted for the default service of one Additional Call per Line on the prime line and the end user has two simultaneous calls, access to another line on the phone will be required to transfer either call on the prime line.

Extensions – meaning that two phones can be on the same line simultaneously simply by picking up the phone – are not available in the VoIP environment.

Barge, a soft key, is available by default. Users can use the Barge feature to access a call on a shared line. For users of shared lines who need the ability to prevent others from joining in on a call, a Private button must be programmed. The Private button will always be the last available button on the telephone set or expansion module. It is activated/deactivated at the user’s discretion and applies to any shared lines on the user’s telephone set.

Softkey features include:

♦ Answer: Answer a call
♦ CallBack: Receive notification when another busy IP phone extension becomes available
♦ Cancel: Cancel an action or exit a screen without applying changes
♦ CFwdALL: Setup/cancel call forwarding (campus and off-campus calls); available only for the prime line, unless accessed via CCM User.
♦ Clear: Delete records or settings
♦ ConfList: View conference participants
Standard Features (cont.)

- Confm: Create a conference call
- Default: Restore settings (including volume) to original factory values
- Delete: Remove characters to the right of the cursor when using EditDial
- Dial: Dial an entered phone number
- EditDial: Edit a number in a call log
- EndCall: Disconnect current call
- Exit: Return to the previous screen
- GPickUp: Answer a call on another extension outside your group
- Hold: Put active call on hold
- Divert: Divert a call immediately to voice mail.
- More: Display additional SoftKeys
- New Call: Make a new call
- Park: Store a call using Call Park
- PickUp: Answer a call on another extension in your group
- Private: Allow/disallow others from viewing or barging calls on a shared line
- Redial: Redial the most recently dialed number
- Remove: Remove a conference participant (select from list)
- Restore: Restore settings (including volume) to previously saved values
- Resume: Resume a call on hold
- RmLstC (Remove Last Caller): Drop the last party added to conference call
- Save: Save the chosen settings
- Search: Search for a directory listing
- Select: Select an item on the screen
- Speed Dial: Direct dial button
- Trnsfer: Transfer a call
- Update: Refresh content and get the latest information
- <<: Delete entered characters
- >>>: Move through entered characters
Additional Features

UTTS IS offers the rental of an IP portable conference unit. Rental rates include the installation of the service and two-day rental of the unit but do not include any toll charges. Installation requires a new telephone number be assigned. If the customer has analog service in a VoIP Building and an analog portable conference unit (Polycom) is needed, UITS IS will rent at the current rates.

Secondary Call Appearances are also available in the VoIP environment. A secondary call appearance is the appearance of a line that is not a primary on any other telephone set.
The VoIP Environment and Analog Service

There are two methods of providing analog services in a VoIP environment – static analog service via a port on a VG248 and mobile analog service via an Analog Terminal Adapter (ATA). The type of service provisioned is dependent upon the building configuration and customer’s needs. Different rates apply for provisioning the two analog services.

**VG248**

Static Analog Service (or simply Analog Service) requires the terminating jack to be wired to a dedicated port on a VG248 that resides in the UITS closet. Any customer ordering analog services in a totally VoIP environment needs to be advised that they will not be able to move the service at their discretion. Furthermore, any jack wired to a VG248 port will not support a VoIP telephone set since the jack in question is “dedicated” for analog service only. To change the dedicated analog jack to a regular VoIP jack or to move an analog device from one jack to another requires the issuance of a service order by IS Telecommunications Services.

**Analog Terminal Adapter (ATA)**

For customers that need the ability to move analog service in a complete VoIP environment at their own discretion, or in buildings which do not have a VG248 provisioned - “Mobile” analog service is offered. Mobile Analog Service will be provisioned with the use of an Analog Terminal Adapter (ATA) 186 which will be installed at the jack. Since ATAs have their own MAC address, all ATAs must be provided by UITS. Customers purchase the ATA device outright, similar to purchasing a telephone set, and are the owners of the equipment.
VoIP Telephones

There are several different telephone sets available for the customer to choose. All IP sets are only offered via purchase. UITS IS does not rent IP end-user equipment. For additional information please contact IS Telecommunications Services at 621-8999.

8 Button VoIP Telephones

- Eight buttons that can be programmed as either a line, speed dial or feature.
- **Touch Screen** LCD display – Provides intuitive access to calling features.
- Four soft keys dynamically present calling options to the user.
- Messages Button - Provides direct access to voice mail.
- Directories Button - Quickly provides directory of Missed, Received, and Placed Calls; access faSt Dials or Personal Address Book; and access the Campus phone directory.
- Settings Button - allows the user to adjust display contrast, volume settings and select unique ring tones.
- Services Button – not developed at this time.
- Help Button - gives users information about the phone's keys, buttons and features.
- Toggle Buttons – allows users to quickly activate or deactivate Headset, Mute or Speaker.
- The scroll toggle bar allows easy movement through the displayed information.
- A volume-control toggle provides easy decibel-level adjustments of the handset, speaker and ringer.
- A foot stand that is adjustable from flat to 60 degrees.
- A speaker on/off button and microphone mute buttons.

* supported, no longer sold
8 Button VoIP Telephones (cont.)

- Busy Lamp Field to monitor the state (in-use or idle) of a call appearance.
- Integrated 10/100/1000 switch (7975G/7971G-GE) and 10/100 switch (7970G)

6 Button VoIP Telephones

- Six buttons that can be programmed as either a line, speed dial or feature.
- LCD display - Provides intuitive access to calling features.
- Four soft keys dynamically present calling options to the user.
- Messages Button - Provides direct access to voice mail.
- Directories Button - Quickly provides directory of Missed, Received, and Placed Calls; access faSt Dials or Personal Address Book; and access the Campus phone directory.
- Settings Button - allows the user to adjust display contrast, volume settings and select unique ring tones.
- Services Button – not developed at this time.
- Help Button - gives users information about the phone's keys, buttons and features.
- Toggle Buttons – allows users to quickly activate or deactivate Headset, Mute or Speaker.
- The scroll toggle bar allows easy movement through the displayed information.
- A volume-control toggle provides easy decibel-level adjustments of the handset, speaker and ringer.
- A foot stand that is adjustable from flat to 60 degrees.
- A speaker on/off button and microphone mute buttons.
- Busy Lamp Field to monitor the state (in-use or idle) of a call appearance (7965G/7961G-GE only).
- Integrated 10/100/1000 switch (7965G/7961G-GE) and 10/100 switch (7960G)

* supported, no longer sold
2 Button VoIP Telephones

- Two buttons that can be programmed as either a line, speed dial, or feature.
- LCD display - Provides intuitive access to calling features.
- Four soft keys dynamically present calling options to the user.
- Messages Button - Provides direct access to voice mail.
- Directories Button - Quickly provides directory of Missed, Received, and Placed Calls; access faSt Dials or Personal Address Book; and access the Campus phone directory.
- Settings Button - allows the user to adjust display contrast, volume settings and select unique ring tones.
- Services Button – not developed at this time.
- Help Button - gives users information about the phone’s keys, buttons and features.
- Toggle Buttons – allows users to quickly activate or deactivate Headset, Mute or Speaker.
- The scroll toggle bar allows easy movement through the displayed information.
- A volume-control toggle provides easy decibel-level adjustments of the handset, speaker and ringer.
- A foot stand that is adjustable from flat to 60 degrees.
- A speaker on/off button and microphone mute buttons.
- Busy Lamp Field to monitor the state (in-use or idle) of a call appearance (7945G/7941G-GE only).
- Integrated 10/100/1000 switch (7945G/7941G-GE) and 10/100 switch (7940G)

* supported, no longer sold

http://uits.arizona.edu/
Basic VoIP Telephones

- LCD display – Provides intuitive access to calling features.
- Four soft keys dynamically present calling options to the user.
- The scroll toggle bar allows easy movement through the displayed information.
- Menu key – Allows users to quickly access directory of Missed, Received, and Placed Calls; access faSt Dials or Personal Address Book; and access the Campus phone directory.
- Hold key – Lighted key provides users a red visual indication that they have placed a call on hold.
- A volume-control toggle provides easy decibel-level adjustments of the handset, speaker and ringer.
- A single-position foot stand that can be removed to allow wall mounting via mounting holes located on the base of the phone.
- Supports call monitor (speaker only, no microphone) – must use handset to be heard.

VoIP Conference Telephones

- Standard features – Includes call hold, call transfer, call release, mute, conference ("ad-hoc" and "meet-me" conferencing), park and pick up.
- Full-duplex operation – Permits natural, two-way conversations without clipping or distortion; the system automatically adapts to changes in the acoustic conditions of the room using state-of-the-art acoustic technology.
- Integrated keypad – Eliminates the need to receive and place calls on a separate telephone.
- 360-degree room coverage – A powerful, digitally-tuned custom speaker and three sensitive microphones provide uniform coverage of small-to-midsize conference rooms or offices.
- Single cable design – Reduces clutter on the tabletop by combining a single cable from the power interface module (PIM) cable with network and power.
- Freedom from special end-user training – Works like a regular telephone.

* supported, no longer sold
- Dynamic Host Configuration Protocol (DHCP) for auto address configuration to the IP network
- The Cisco Unified IP Conference Station 7936 uses a single 10/100BaseTx Ethernet LAN connection to the network via a RJ-45 cable interface.
- Convenient volume control buttons
- Five user-adjustable ring tones
- Comfort noise generation and voice activity detection
- Local 20 entry directory
- Extension microphones available for purchase.
- LCD display – Provides intuitive access to calling features.

**Expansion Module**

Currently available for 7971G-GE, 7970G, 7961G-GE and 7960G. (Similar product for 7975G & 7965G under testing.)
- Extend capabilities of 8 and 6 Button VoIP Telephone Sets with additional buttons and an LCD display.
- Additional buttons can be programmed as a line or as a speed dial.
- Illuminated Buttons – Change color to indicate line status; Off/Dark = Line Available; Green, steady = Line in use by you; Red, steady = Line in use by someone else; Amber, flashing = Line ringing; Green, flashing = You have the call on hold; Red, flashing = Someone else has the call on hold.
- Foot stand – Single foot stand and double foot stands are available.

**7914:**
- Each expansion module adds 14 buttons to the existing 7960 or 7970 set.
- Up to two 7914 expansion modules can be used per telephone set.
End User Documentation

Documentation including instructions on configuring your PC for auto-negotiation, training material on the VoIP telephone sets, and training material on Cisco Call Manager (CCM) User are all available at http://uits.arizona.edu/index.php?id=691.