



Telecenter[®]
by *Rauland*



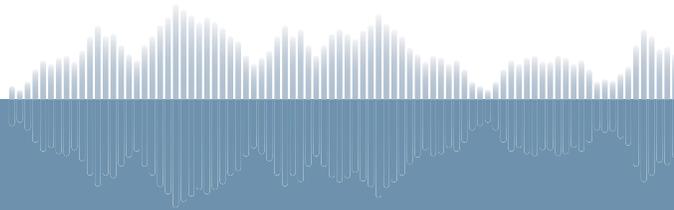
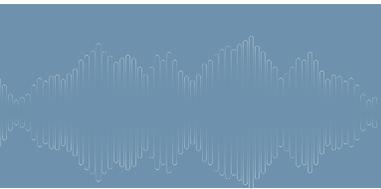
Application Note

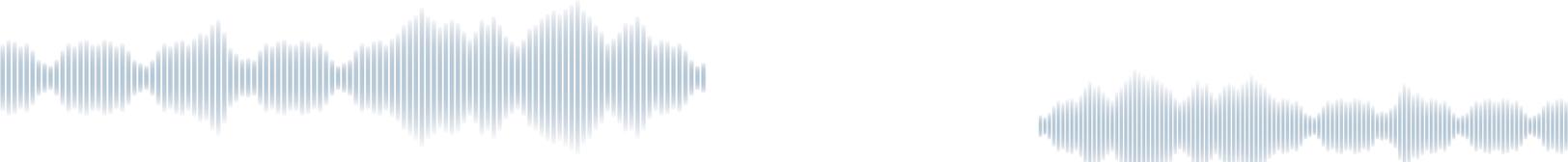
EAN 0037

Telecenter U[®], Campus Edition SIP Integration to an Analog PBX



Connect **Your** Universe[®]





Overview

While Telecenter U is designed to interface via SIP to the school's phone system, some schools do not have SIP enabled PBX systems. These schools would still like to benefit from interfacing Telecenter U to their phone system. By using the Patton SmartNode gateway outlined in this application note, schools can connect Telecenter U to their PBX and benefit from the following features from any PBX connected phone:

- Receive call-ins, with call priority and room number displayed via caller ID
- Answer call-ins
- Intercom into individual classrooms
- Live page into zones as well as school wide all page.
- Initiate emergencies, sequences and prerecorded audio

This Application Note outlines the method and equipment necessary to allow access to TCU from a school's analog C.O. line (FXO) using the TCU's SIP interface.

TCU allows phone access through a SIP trunk line connection. When a school has only analog phone lines, they must convert the analog line to a SIP (digital) line. This conversion is handled by a Patton SmartNode 4520 Series Gateway Router. The SmartNode provides standards compliant Voice over IP (VoIP) conversion in accordance with SIP and H.323 protocols. The SmartNode 4520 Series are available in various port configurations. Rauland has tested this application using the Patton SN4524/2JS2JO/UI Gateway Router which includes two (2) FXS ports and two (2) FXO ports. This SmartNode configuration provides a school with up to two analog communication lines into TCU. All FXS ports allow connection to legacy (analog) tie lines or PBX and provide dial tone, caller ID, and ringing. The SmartNode 4520 Series can be configured with up to eight (8) FXS ports.

In this Application Note, the analog C.O. FXO port is connected to the SmartNode Gateway Router's FXS port. See Figures 1 and 2 for process flow details. For additional information and installation details of the Patton SmartNode 4524, visit the Patton website at this link: <http://www.patton.com/voip-router/sn4520/>.

Note: Check-in information does not appear on the PBX phone's display. At least one TCC2044 console should be present in any school equipped with TCU.

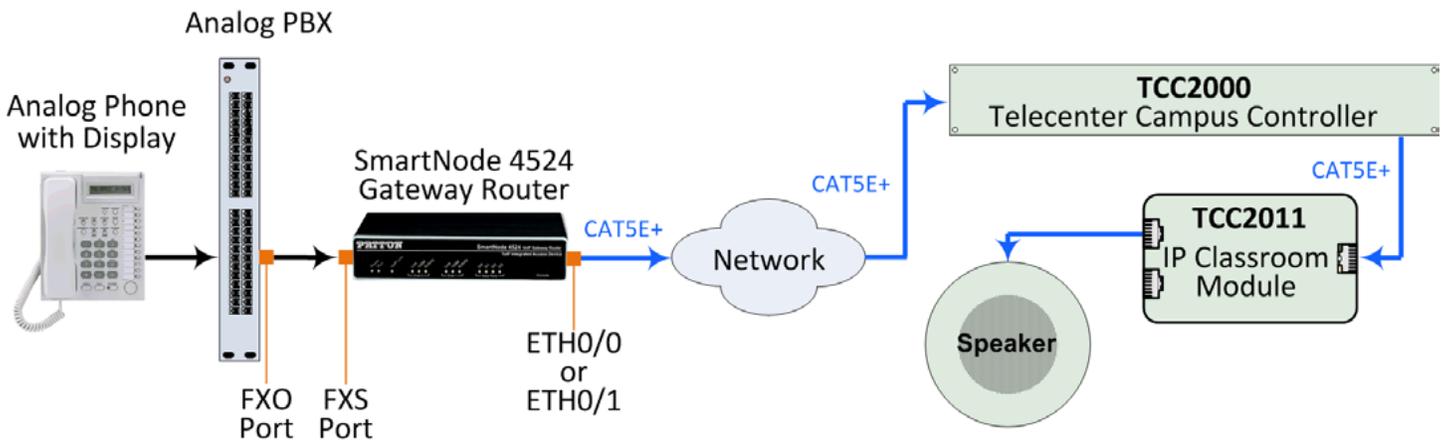


Figure 1 – PBX Phone Dial Action, i.e., Emergency Page

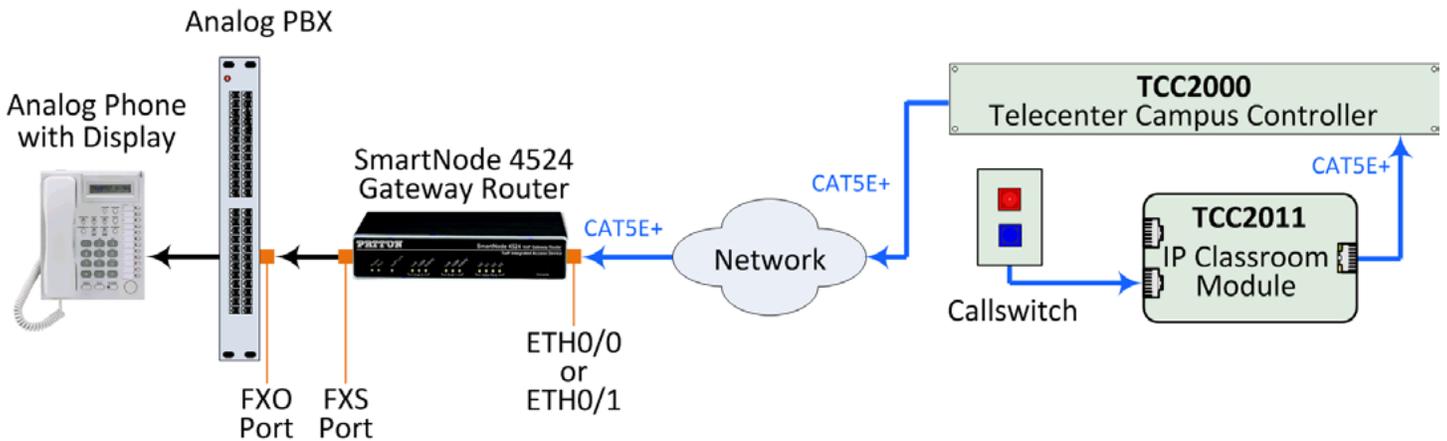
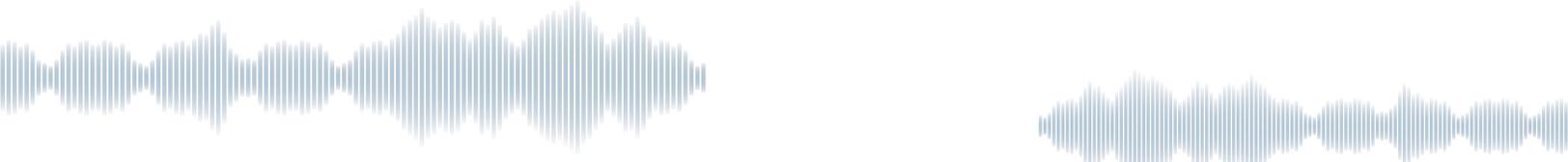


Figure 2 – Telecenter U Classroom Call-in to PBX Phone

Required Material for Telecenter U SIP to Analog Line Connection

- TCC2000 Campus Controller
- TCC2044 Administrative Console
- Patton SmartNode 4524 VoIP Gateway
- Laptop or PC with access to the network connected to the SmartNode 4524 Gateway
- SmartNode Configuration File *TCU-CE_to_Patton_AA_r141121.CFG* available on the Rauland [Customer Connection](#)
- Patton SmartNode User Guide (for installation guidelines, available online and downloadable from the [Patton website](#))
- An available FXO port on the school's analog PBX
- Appropriate RJ11 and RJ45 patch cables for connecting hardware devices
- Analog PBX connected phone



Patton SmartNode 4524 Gateway Router Configuration

Initial setup of the SmartNode Gateway requires loading a configuration. A sample configuration file, *TCU-CE_to_Patton_AA_r141121.CFG* is provided at the end of this Application Note, and is also available on the Customer Connection using the following path:

File Center | Education | Telecenter U | Software

Although your configuration will be very similar to the provided sample, you must modify the configuration to match your network addressing scheme. Review the fully commented sample configuration. The .CFG file is a plain text file and can be viewed and modified using a basic text editor program such as Notepad.

Required Network Information:

- IP address and subnet mask to be used for the SmartNode
- Default gateway for the SmartNode
- Campus Controller IP address
- NTP server IP address

To connect the SIP interface of TCU to an analog PBX, the SmartNode configuration file must be updated with your information and loaded onto the SmartNode Gateway.

1. Power up the SmartNode – plug in the SmartNode power supply to local mains power. Wait until the Run LED stops blinking and stays on constantly.
2. Connect an Ethernet cable between your computer and the ETH1 port on the SmartNode. The SmartNode 4524 is equipped with auto-crossover Ethernet ports, so a crossover cable is not necessary for this connection.
3. Configure the wired network connection on your computer to have an IP address of 192.168.1.10 and a subnet mask of 255.255.255.0.

Required Network Information:

Modify the configuration file with your specific IP addresses. Be sure to follow the guidelines stated in the comments of the configuration file and do not modify the file beyond those that are detailed here or in the comments of the .cfg file.

If your room extensions are other than 3 digits long, you must modify the routing table rule to match the length of your room extensions. Using the example routing rule, your room extensions and the dial numbers of any configured dial actions must all be the same length. While it is possible to modify the configuration to support different length extensions, it is beyond the scope of this document. For details, consult the Patton SmartWare Software Configuration Guide available from the Patton website.

Once you have updated the configuration file for your specific application, save the modified file with a new filename, e.g. TCutoPatton.cfg.

1. Using an internet browser, navigate to the 192.168.1.1. To access the SmartNode, enter the credentials below (SmartNode factory defaults) and click the *Log In*.

- Login - administrator
- Password - <Enter> (empty, hit your Enter key after the Password prompt)

For security purposes, it is recommended that the default login name and password of the SmartNode Gateway Router should be changed upon installation.

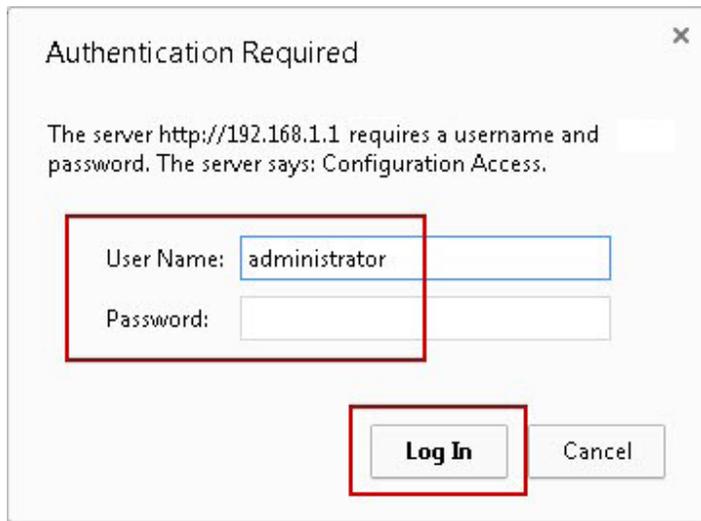


Figure 3 – SmartNode Login

2. After logging in, click on the Import/Export link in the upper left hand portion of the screen.

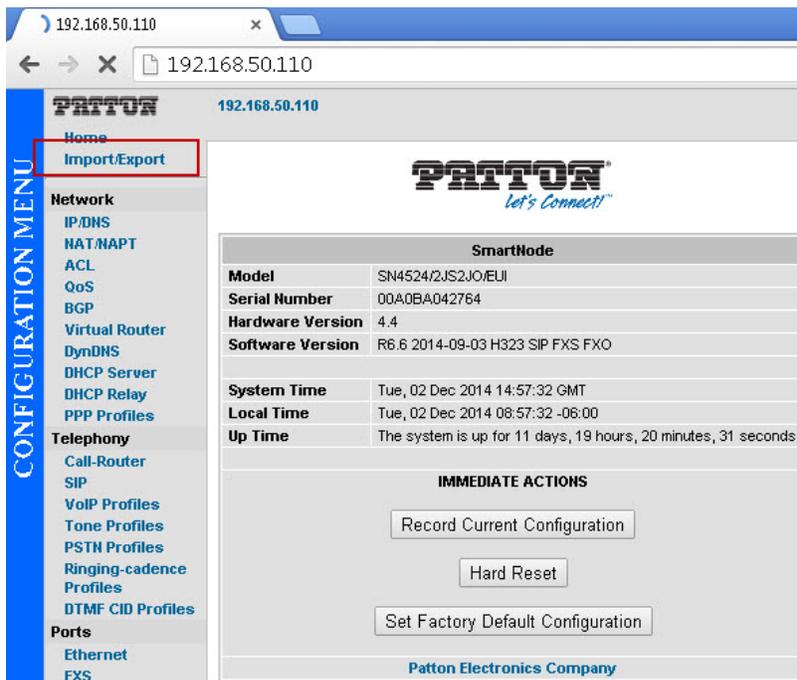


Figure 4 – SmartNode Import/Export New Configuration

3. Click the Import Configuration tab



Figure 5 – SmartNode Import

4. Click the **Choose File** button and navigate to the folder where you saved your updated configuration file (TCUtoPatton.cfg in this example), click on it and the click the **Open** button.

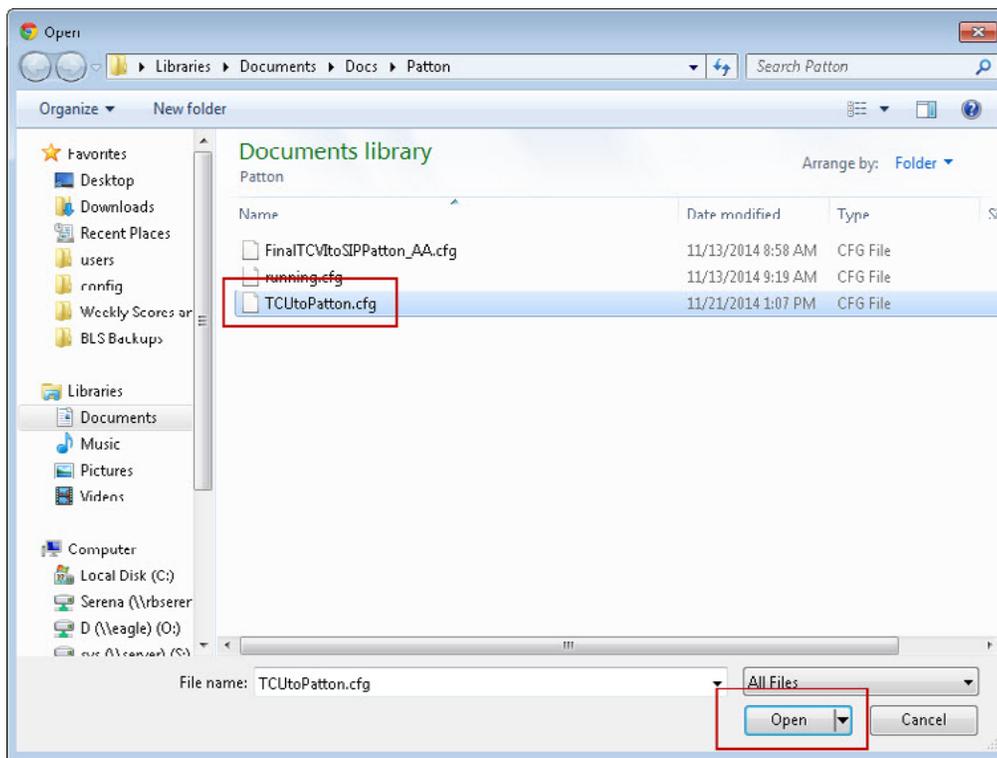


Figure 6 – SmartNode Find New Configuration

5. Click the **Import** button

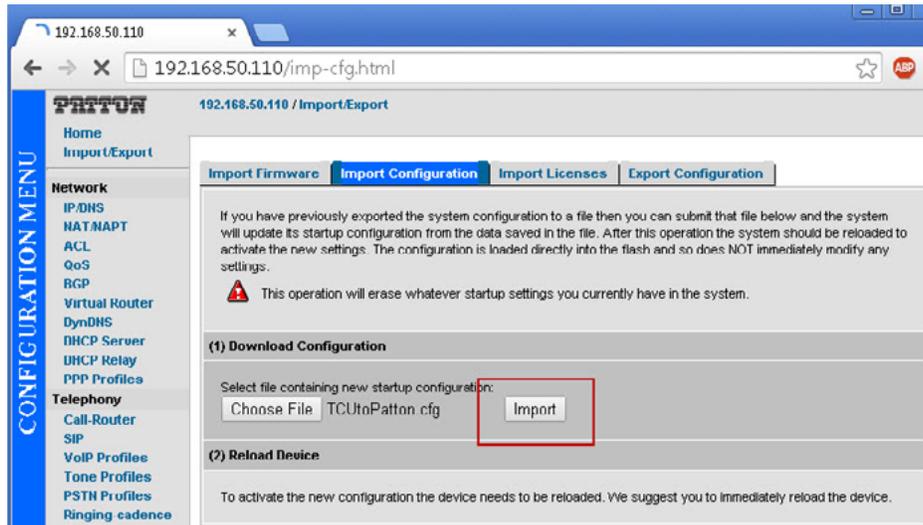


Figure 7 – SmartNode Import

6. Verify the message “The startup configuration has been imported successfully.” appears as shown below and click the **Reload** button.

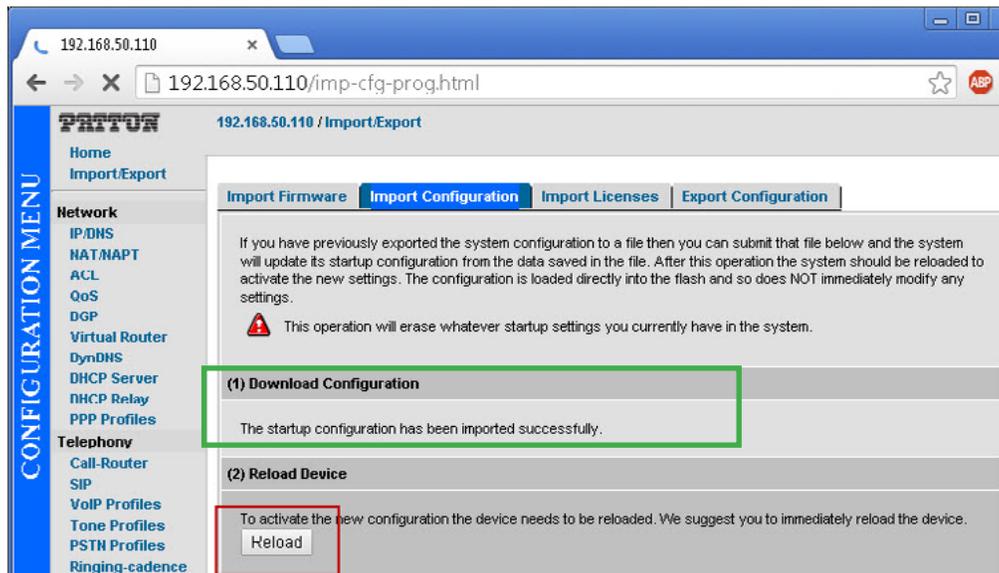


Figure 8 – SmartNode Reload

7. Click the **Reload** button under Reload Device to load your new configuration into the SmartNode. Reloading the SmartNode configuration may take an extended period. Be patient while waiting for the LINK led to light on the front panel of the SmartNode Gateway.

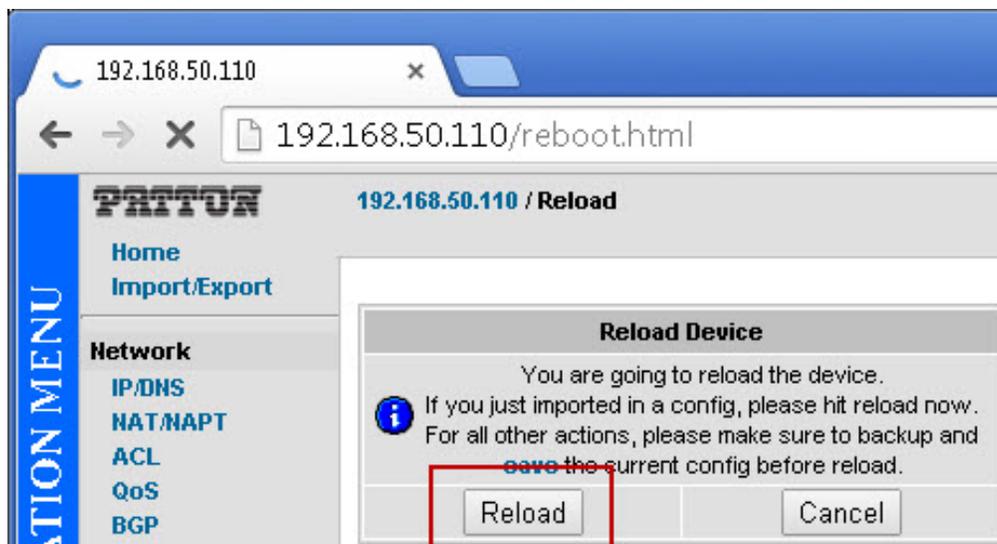


Figure 9 – SmartNode Reboot

Make Network Connections

Disconnect the Ethernet cable between your computer and the SmartNode and restore your wired network connection settings.

Put the SmartNode in place and connect it to its intended network port.

Ensure that you are able to ping the SmartNode (using the IP address you specified in the configuration file) from a computer connected to the network.

Connect Analog Equipment

Connect the analog C.O. lines (FXO) to the SmartNode FXS ports. All analog connections on the SmartNode use standard RJ11 connectors.

The provided SmartNode configuration will hunt FXS ports in descending order (FXS1 then FXS0) when an inbound SIP call is received. To reduce the possibility of glare, connect the analog C.O. lines in such a way that the FXS ports are hunted in ascending order for outbound calls.

TCU Configuration

TCU annunciates call-ins via SIP by assigning a call-in coverage to a “SIP Phone”. When integrated directly with a SIP PBX, these SIP Phones would correspond with actual SIP endpoints on the PBX. In this application, they are simply placeholders for the analog lines on the SmartNode.

Because TCU will only annunciate a single call-in at a time for any given SIP Phone, it is recommended that a SIP Phone is configured for each FXS port used on the SmartNode with the rooms with call switches divided equally among the SIP Phones. Thus, if there are 100 rooms with call switches and 4 FXS ports in use on the SmartNode, then each SIP Phone should be configured to cover roughly 25 rooms. The coverages may be adjusted based on usage. For example, if there are certain rooms with higher than average call-in volume, it may be desirable to separate them such that only one high-volume room is contained in each coverage.

Since the SmartNode itself has no SIP extensions and simply routes any SIP call received regardless of called party number, the SIP Phone extensions chosen are irrelevant for this application. However, it is recommended that they be contiguous and be easily distinguishable (i.e. don't attempt to fit them into your dialing plan). If you have other SIP integrations, bear in mind that SIP Phone extensions must be unique.

Configure SIP Trunk

Using the TCU Web interface, navigate to the following page: *Setup* → *SIP Trunk*, and then click the **Add New SIP Trunk** button below the SIP trunks window.

The screenshot shows the Telecenter U by Rauland web interface. At the top right, there are links for Home, Help, Doc, and Logout. Below the logo, it says "Welcome Larry". A navigation menu includes "Everyday Communication", "Calendar", "Emergencies", "Tools", "Shortcuts", and "Setup". The breadcrumb trail is "Home > Setup > SIP Trunk". The main form has three fields: "SIP Extension Range" with input "100" and "200", "SIP PBX FQDN/IP" with input "38.102.68.3", and "Description" with input "Hudson H.S. SIP trunk interface". Red arrows point to the first two fields with labels: "←Range of phone extensions" and "←IP address of SmartNode Gateway". A "Save" button is at the bottom right of the form. The footer shows "Version U3.1 ©2014 Rauland-Borg Corporation" and the Rauland logo.

Figure 10 – SIP Trunk Configuration

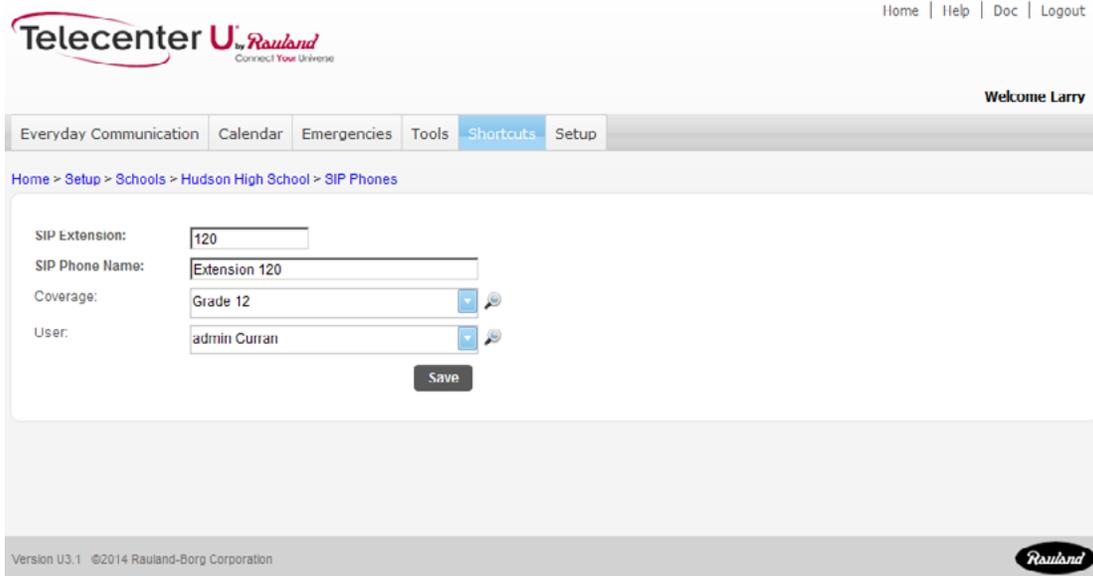
1. Enter the SIP Extension Range. This is the range of phone extensions that will be routed to the SmartNode. They can be from 1 to 12 digits each.
2. Enter the IP address of the SmartNode Gateway that was specified in the configuration.
3. Enter a description of the SIP trunk. This is optional, but recommended.
4. Click the **Save** button to save your SIP trunk configuration

Configure Coverages

Configure a call-in coverage for each FXS port used on the SmartNode as described above. Coverages are configured on the *Setup* → *Schools* → *Configure (after your chosen school)* → *Coverages*, screen. For additional details about Coverages, see KI-2277, the TCU System Programming & Administration Manual.

Configure SIP Phones

Using the TCU Web interface, browse to the following page: *Setup* → *Schools* → *Configure* (after your chosen school) → *SIP Phones*.



The screenshot shows the Telecenter U by Rauland web interface. At the top, there is a navigation bar with links for Home, Help, Doc, and Logout. Below this, a welcome message reads "Welcome Larry". A secondary navigation bar contains tabs for Everyday Communication, Calendar, Emergencies, Tools, Shortcuts, and Setup, with Setup being the active tab. The main content area displays a breadcrumb trail: Home > Setup > Schools > Hudson High School > SIP Phones. Below the breadcrumb is a form for configuring a SIP phone. The form contains four fields: SIP Extension (text input with value 120), SIP Phone Name (text input with value Extension 120), Coverage (dropdown menu with value Grade 12), and User (dropdown menu with value admin Curran). A Save button is located below the User field. At the bottom of the page, there is a footer with the text "Version U3.1 ©2014 Rauland-Borg Corporation" and the Rauland logo.

Figure 11 – Telecenter U Phone Configuration

1. Click the **Add New SIP Phone** button below the SIP Phones window
2. Enter one of your SIP extensions. The SIP phone extension must be unique to the school and contain from two to six digits. This example uses 3-digit SIP phone extensions.
3. Enter the SIP phone name. The name is used for identification of the SIP phone through the TCU configuration interface (Web GUI). The SIP phone name can be up to 20 alphanumeric characters and must be unique from other SIP phones in the same school.
4. Select the Coverage for this SIP Phone from the drop-down list. There should be a coverage for each SIP Phone as described above.
5. Click the **Save** button to save your SIP phone configuration.
6. Repeat steps 1-5 to add as many SIP Phones as FXS ports you plan to use on the SmartNode.

Configure Dial Actions

To perform actions other than intercom (e.g. Paging or Emergency initiation), Dial Actions must be configured. Dial Actions are configured on the *Setup* → *Schools* → *Configure (after your chosen school)* → *Dial Actions* screen.

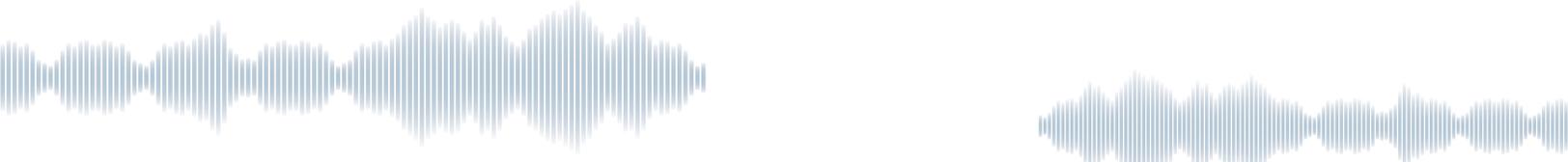
The Dial Action configured below initiates a Lockdown Emergency when an analog phone connected to TCU dials 992. Additional details for Dial Actions are available in the TCU System Programming & Administration Manual, KI-2277.

The screenshot shows the 'Telecenter U. of Rauland' web interface. The top navigation bar includes 'Home | Help | Doc | Logout' and a 'Welcome Larry' message. The main menu has tabs for 'Everyday Communication', 'Calendar', 'Emergencies', 'Tools', 'Shortcuts', and 'Setup'. The breadcrumb trail is 'Home > Setup > Schools > Hudson High School > Dial Actions'. The configuration form includes:

- Dial Action Extension:
- Description:
- Action:
 - All Page
 - Emergency All Page
 - Zone Page
 - Terminate All Pre-recorded Audio
 - Pre-recorded Audio
 - Cancel All Call-Ins In This School
 - Terminate All Sequences
 - All Clear
 - Set Swing
 - Clear Swing
 - Toggle Swing
 - Run Sequence
 - Run Emergency
 - Toggle Swing
- Duration: s
- Target:
- Attachment:
- Save button

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Figure 12 – Dial Action Configuration Example



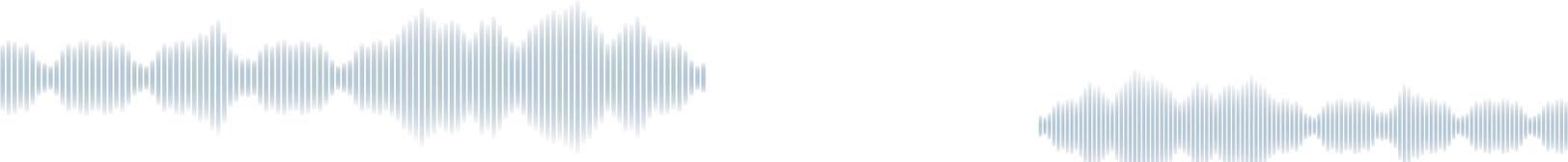
Testing

It is recommended to test the SmartNode isolated from any PBX equipment by simply plugging analog caller-id phones into the FXS ports. This will allow the verification of TCU and SmartNode programming before PBX integration is attempted. At a minimum, ensure the following scenarios are tested:

- Intercom call to a single room
- Multiple, simultaneous intercom calls
- Paging dial action initiation (if desired)
- Emergency dial action initiation (if desired)
- Call-in placement from a single room
- Call-in placement from multiple rooms in different coverages (all call-ins should annunciate)
- Call-in placement from multiple rooms in the same coverage (only the highest priority call-in should annunciate)

Summary

The Telecenter U, Campus Edition to analog CO line integration provides for much broader sales opportunities. Now all schools and districts can be equipped with the latest technology to address everyday communications as well as critical emergencies. The addition of a Patton SmartNode Gateway Router is an inexpensive add-on to TCU that facilitates enhanced safety and emergency response from any PBX system.



The following SmartNode configuration file, *TCU-CE_to_Patton_AA_r141121.CFG*, is also available on the Rauland [Customer Connection](#). The information below is an example of the SmartNode CFG file and must be configured per installation details mentioned above.

```
#-----#
#
# SN4524/2JS2JO/EUI #
# R6.6 2014-09-03 H323 SIP FXS FXO #
# 2014-11-21T16:51:20 #
# SN/00A0BA042764 #
# Telecenter U Campus to Analog Config file Patton SN452X/JO/EUI #
# (C)2014 Rauland-Borg Corporation #
# See all comments below that start with # #
#-----#

#
# Digits dialed on the FXS interfaces will be matched based on the
# routing table rule described below and converted to a SIP call to
# the Telecenter U Campus controller.
#
# SIP calls received from the campus controller will be routed out the FXS
# interfaces in descending order (FXS 0/1 then FXS 0/0).
#
# The analog PBX should be connected via its FXO ports to the Patton's FXS
# ports. The analog PBX should be configured to hunt in ascending order
# (FXS 0/0 then FXS 0/1) to minimize glare.
#
# The campus controller should be configured with as many SIP phones as
# there are tie lines. Each SIP phone should be configured to cover roughly
# the inverse of the number of tie lines of the endpoints (e.g. if there
# are 2 tie lines, each SIP phone should cover 1/2 of the endpoints). Only
# 1 call-in will be annunciated per configured SIP phone at any given time.
# A SIP trunk must be configured that includes all the SIP phone extensions
# and points to the Patton's IP address configured below. The SIP phone
# extensions are immaterial.
#
# NOTE: The Patton will not truncate the caller ID name received from a
# SIP call when routing the call out the FXS interface. If the total
# length of the caller ID name (<priority> - <room name>) exceeds 15
# characters (the PSTN maximum length), some PBXs may not be able to
# process and display the caller ID name.
#
# ----->>>>> ALL LINES THAT REQUIRE AN UPDATE ARE PRECEDED BY THE
# ----->>>>> SYMBOLS SHOWN TO THE LEFT (----->>>>>)
#

cli version 3.20
dns-relay
webserver port 80 language en
```



```
#
# Local Clock Settings
#
# sntp-client server primary x.x.x.x port 123 version 4
#
# Where x.x.x.x is the IP address of the SNTP server to sync time to.
#
# clock local default-offset (+)hh:mm
#
# Where (+/-)hh:mm is the local offset from GMT in hours and minutes.
#
sntp-client

# ----->>>>>> Be sure to use the same time source as the TCC2000
# ----->>>>>> and the TCU District Server on the line below

sntp-client server primary 192.168.50.21 port 123 version 4
clock local default-offset -06:00

system

  ic voice 0
    low-bitrate-codec g729

profile ppp default

profile call-progress-tone defaultDialtone
play 1 1000 350 -13 440 -13

profile tone-set default

profile voip default
codec 1 g711alaw64k rx-length 20 tx-length 20
codec 2 g711ulaw64k rx-length 20 tx-length 20

profile pstn default

profile ringing-cadence default
play 1 1000
pause 2 4000

profile sip default
no autonomous-transitioning

profile aaa default
method 1 local
method 2 none

#
# IP Address of ETH 0/0 Interface
#
```



```
context ip router
interface eth0
#
# ipaddress x.x.x.x yyyy
#
# Where x.x.x.x is the IP address and yyyy is the subnet mask
# of the SmartNode Gateway
#
# ----->>>>>>
# ----->>>>>>

ipaddress 192.168.50.110 255.255.248.0
tcp adjust-mss rx mtu
tcp adjust-mss tx mtu
```

```
context ip router
#
# Default Gateway
#
# route 0.0.0.0 0.0.0.0 x.x.x.x 0
#
# Where x.x.x.x is the IP address of the default gateway (router).
#
# ----->>>>>>
# ----->>>>>>

route 0.0.0.0 0.0.0.0 192.168.50.2 0
```

```
context cs switch
#
# Routing Table for TCU Extensions
#
# route ... dest-interface IF_SIP_0
#
# Where ... is a regular expression that will match exactly 3 digits. Adjust
# the number of dots to match the length of TCU Campus extensions.
# ALL PHONE EXTENSIONS, DIAL ACTIONS AND ROOM NUMBERS MUST
# BE THE SAME NUMBER OF DIGITS
#
routing-table called-e164 ROUTE_TO_SIP
#
# ----->>>>>>
# ----->>>>>>

route ... dest-interface IF_SIP_0

interface sip IF_SIP_0
bind context sip-gateway GW_SIP_0
route call dest-service HG_FXS_DESC
```



```
#
# TCU Campus Controller SIP Interface
#
# remote x.x.x.x
#
# Where x.x.x.x is the IP address of the Campus controller.
#
# ----->>>>>>
# ----->>>>>>

remote 192.168.50.118
address-translation incoming-call calling-name from-header

interface fxs IF_FXS_0
route call dest-table ROUTE_TO_SIP
no call-waiting
caller-id-presentation mid-ring
caller-id-format bell

interface fxs IF_FXS_1
route call dest-table ROUTE_TO_SIP
no call-waiting
caller-id-presentation mid-ring
caller-id-format bell

service hunt-group HG_FXS_DESC
drop-cause normal-unspecified
drop-cause no-circuit-channel-available
drop-cause network-out-of-order
drop-cause temporary-failure
drop-cause switching-equipment-congestion
drop-cause access-info-discarded
drop-cause circuit-channel-not-available
drop-cause resources-unavailable
drop-cause user-busy
route call 1 dest-interface IF_FXS_1
route call 2 dest-interface IF_FXS_0

context cs switch
no shutdown

context sip-gateway GW_SIP_0

interface LAN
bind interface eth0 context router port 5060

context sip-gateway GW_SIP_0
no shutdown

port ethernet 0 0
medium auto
encapsulation ip
```



```
bind interface eth0 router  
no shutdown
```

```
port ethernet 0 1  
medium auto  
no encapsulation  
shutdown
```

```
port fxs 0 0  
encapsulation cc-fxs  
bind interface IF_FXS_0 switch  
no shutdown
```

```
port fxs 0 1  
encapsulation cc-fxs  
bind interface IF_FXS_1 switch  
no shutdown
```

```
port fxo 0 0  
shutdown
```

```
port fxo 0 1  
shutdown
```



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