

Configuring the Grandstream UCM6202 for use with TopView[®] Voice Notification

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Recent updates:

- March 26, 2021
 - Upgrade firmware instructions added
 - Instructions updated for UI changes in newer firmware
 - New recommended setting for concurrent registrations of extension
- January 31, 2021: added steps to verify analog line, warning about line filters

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Introduction

Voice Notification in TopView

TopView Voice Notification allows users to make phone calls upon events such as alarms. The phone call can speak a message to the user and the user can optionally acknowledge the event using their phone keypad (default is the # key).

Two methods are available for making voice phone calls in TopView.

- **TAPI** (telephony API) requires a hardware device and an analog phone line. The hardware device is a TAPI device or voice modem connected to the TopView machine and is either internal or external (serial/USB) to the TopView machine. Drivers from the hardware device vendor must be installed before TopView can communicate to the hardware device.
- **VOIP** requires a VOIP Server (SIP Server). TopView can communicate to the SIP Server over the network to make phone calls. The connection from TopView to the SIP Server is a network connection. The SIP Server is responsible for making the phone call. The SIP Server may be internal on the company LAN (if part of a company phone PBX) or may be a hosted SIP Server through a paid service (requires internet access from the TopView machine).

Because we are phasing out support for TAPI we recommend that users implement voice notification using VOIP.

Options for using VOIP in TopView

- **Use your internal phone system:** If your internal phone system supports VOIP (it has a SIP Server) and you can access the SIP Server from the TopView machine to make phone calls.
- **Hosted SIP Server:** If your internal phone system cannot be used and the TopView machine can access a hosted SIP Server. This option requires network access from TopView to the hosted SIP Server on the Internet. You will also need to get an account with a SIP Server host.
- **Host your own SIP Server:** use an IP PBX to host your own SIP Server. The IP PBX can be connected to analog phone lines for outgoing calls and will also allow IP and soft phones (software phones) on the same network to receive calls from TopView.

The Grandstream UCM6202 is an IP PBX

How can TopView use the UCM6202?

The Grandstream UCM6202 is an “IP PBX”. For TopView users it provides a SIP Server and connection to analog phone line(s) so VOIP can be used to make voice notification calls.

If an analog phone line is connected to the UCM6202, TopView can make VOIP calls that go out over this analog line. Users on the same network as TopView can also implement IP phones (hardware) and soft(ware) phones (running on smartphones or desktop computers) to receive voice notification calls from TopView.

UCM6202 uses for TopView customers:

- Provide a bridge for TopView VOIP support and an existing analog phone line
 - Replace an existing TAPI solution that uses an analog phone line to make calls
 - Implement VOIP voice notification if your internal phone system or hosted SIP server options cannot be used.
- Provide a SIP Server for implementing TopView voice notification to IP phones and soft phones running on smartphones and computers on the same LAN as TopView.

Configuration Overview

1. Configure UCM6202 for your network
2. Use a browser to connect to UCM6202 web configuration tool
3. Make changes to the UCM network settings to communicate with your network
4. Configure extensions (for TopView and optionally for IP Phones and softphones)
5. Connect and configure the analog line (analog trunk)
6. Configure dialing sequence to access the analog line from TopView and other VOIP phones (outbound rule)
7. Make test calls

The UCM6202 (UCM) LCD display

The UCM6202 has an LCD display and keypad that you can use to view and edit some of the configuration details for the device. The current settings can be viewed using this display and the buttons to the right of the display and some settings can also be changed. This document will use the web configuration utility to make configuration changes.

Configuring the UCM6202 (UCM) on your network

The UCM can has 2 Ethernet ports labeled WAN and LAN.

The UCM can operate as a router or a switch. By default the UCM is operating as a Router.

Router defaults:

- WAN port: if connected the UCM is a DHCP client and will try to use your DHCP server to get an IP address. If DHCP is successful you will see the IP address displayed on the device LCD display.
- LAN port: has a static IP address of 192.168.2.1 and is operating as a DHCP Server. This can be displayed through the menu of the LCD display if you go into the network settings menu.

You can connect to the UCM using the LAN or WAN port.

- WAN: If you have an existing network with a DHCP Server you can connect the UCM to your network using the WAN port. The UCM IP address of the device is displayed on the device. Use this IP address below.
- LAN: You can connect a computer directly to the UCM with an Ethernet cable connected to the LAN port. The UCM IP address is 192.168.2.1. The computer's ethernet adapter should be set to use DHCP so it will receive an IP address on the 192.168.2.x network. It will most likely get the address 192.168.2.100.

Open a web browser and enter the URL of the UCM as

<http://ipaddress>

Example: <http://192.168.2.1> if using the LAN port

Web Configuration

For this example we are performing our initial connection to the UCM over the LAN port (192.168.2.1). If you are using the WAN port you should use the IP address shown on the LCD display.

Log into the device

Open your web browser and enter the address

<http://192.168.2.1>

Log into the device:

Note: depending on when you purchased your device, it may or may not have a unique username and password. Check the bottom of the device to see if there is a sticker with your username and password.

If you have a sticker on the bottom of the device with the username and password:

- Username: *name on sticker*
- Password: *password on sticker*

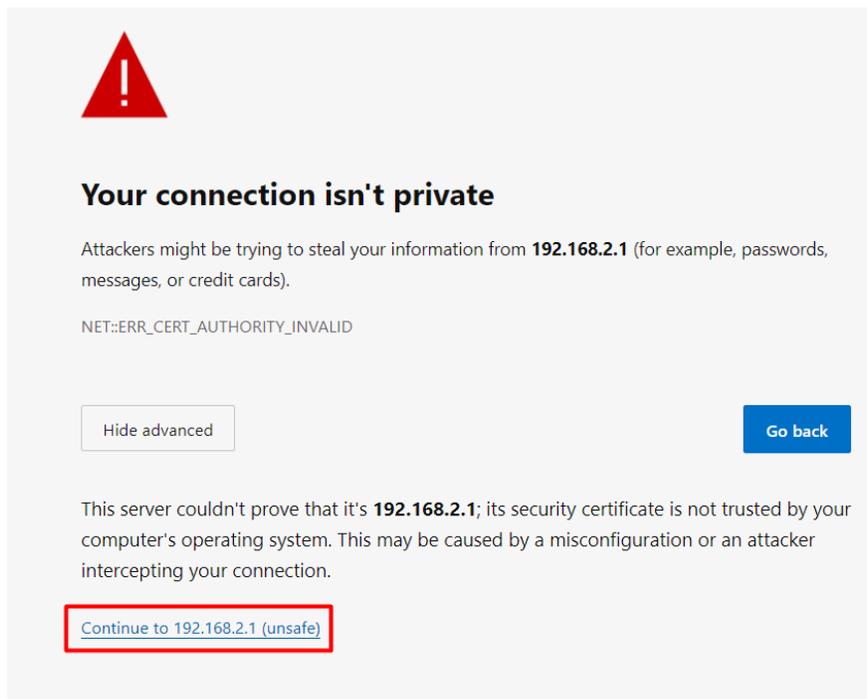
If you do not have a sticker on the bottom of the device with the username and password:

- Username: admin
- Password: admin

If the setup wizard starts you can [Quit] for now.

Chrome (connection warning)

If you receive a warning in Chrome/Edge "connection is not private" click on Advanced and Proceed.



Configure the network settings

- [System Settings] from the left menu
- Then select [Network Settings]

The goal of this step is to ensure that the UCM has a static IP address that can be accessed from the TopView machine.

Router or Switch?

We expect that most customers will want to add the UCM to an existing network. Therefore they will want to change the Network Method from Route to Switch. Both methods will work as long as TopView can access the UCM using a known IP address.

Our example will change the UCM to act as a switch on an existing network. The existing network has IP addresses 172.16.1.X. We will set the UCM to have a static IP address of **172.16.1.20**

Basic Settings (example):

- Method: change to Switch.
 - **In Switch mode the UCM will have a single IP address and you can use the LAN or WAN ports on the rear of the UCM to access the UCM over the network.**
- LAN IP Method: change to Static and enter the desired IP address, subnet, ...

The screenshot shows a network configuration interface. At the top, there are two input fields: 'Method:' with a dropdown menu set to 'Switch', and 'MTU:' with a text box containing '1500'. Below these is a tabbed interface with 'IPv4 Address' selected. Under the 'IPv4 Address' tab, there is a 'Preferred DNS Server:' field. Below that is a section titled 'LAN' containing several fields: 'IP Method:' with a dropdown set to 'Static'; '* IP Address:' with '172.16.1.20'; '* Subnet Mask:' with '255.255.255.0'; '* Gateway IP:' with '172.16.1.1'; '* DNS Server 1:' with '172.16.1.1'; 'DNS Server 2:' (empty); 'Layer 2 QoS 802.1Q/VLAN Tag:' with '0'; and 'Layer 2 QoS 802.1p Priority' with '0'.

- Save the settings and restart the device

- From a computer on the same network as the UCM, access the web configuration using the static IP address (172.16.1.20 in our example)

Update the firmware

- Check the firmware version of your Grandstream device
 - [System Status] from left menu, then select [System Information]
 - Version information: Base
- Check the [latest firmware version](#) from Grandstream.

If your device firmware is not current:

- Download the firmware file (zip)
 - Unzip the .bin file from the downloaded zip
 - From the Grandstream web configuration
 - [Maintenance] from the left menu then [Upgrade]
 - Firmware file path: point to the .bin file
- WARNING: the firmware upgrade will begin immediately

Time zone (optional)

- [System Settings] from left menu
- Then select [Time settings]

Set the time zone to your time zone

SIP Server

- [PBX Settings] from left menu
- Then select [SIP Settings]

The SIP Server is the VOIP Server accessed by TopView, soft phones, and IP phones.

In our example we are not making any changes to the SIP Server. The SIP server will be available on the static IP address of the UCM (**172.16.1.20**) and we will leave the default port as 5060.

- Port: the default port is 5060 and should be OK for most users

General

Misc

Session Timer

TCP/TLS

NAT

- * Realm for Digest Authentication:
- * Bind UDP Port:
- * Bind IPv4 address:
- * Bind IPv6 address:
- Allow Guest Calls:
- Allow Transfer:
- MWI From Header:
- Enable Diversion Header:

Extensions

- [Extension/Trunk] from the left menu
- Then select [Extensions]

Extensions allow each VOIP client (TopView, soft phone, IP phone ...) to connect to the UCM's SIP Server.

An extension for a SIP server is similar to an email address for an email server. VOIP clients will use an extension and password to log into the SIP server to make phone calls. Soft phones and IP phones can also receive phone calls.

For our example we will create 10 SIP extensions from 1001 to 1010 with the password abc5.

Click the [Add] button in the top left corner. You should be on the [Basic Settings] tab

- Extension type: SIP Extension
- Select Add Method: Batch
- Create Number: 10
- Extension: 1001
- Permission: Internal
- SIP/IAX password: **abc5**
- Click [Save] (or for older firmware [Create users])
- Click [Apply changes] at the top of the screen to apply these changes

Extensions

<input type="checkbox"/>	STATUS	PRESENCE STATUS	EXTENSION	NAME	TYPE	IP AND PORT	EMAIL ST...	OPTIONS
<input type="checkbox"/>	● Unavailable	Available	1001		SIP(WebRTC)	--		
<input type="checkbox"/>	● Unavailable	Available	1002		SIP(WebRTC)	--		
<input type="checkbox"/>	● Unavailable	Available	1003		SIP(WebRTC)	--		
<input type="checkbox"/>	● Unavailable	Available	1004		SIP(WebRTC)	--		
<input type="checkbox"/>	● Unavailable	Available	1005		SIP(WebRTC)	--		
<input type="checkbox"/>	● Unavailable	Available	1006		SIP(WebRTC)	--		
<input type="checkbox"/>	● Unavailable	Available	1007		SIP(WebRTC)	--		
<input type="checkbox"/>	● Unavailable	Available	1008		SIP(WebRTC)	--		
<input type="checkbox"/>	● Unavailable	Available	1009		SIP(WebRTC)	--		
<input type="checkbox"/>	● Unavailable	Available	1010		SIP(WebRTC)	--		

Verify extension properties, set concurrent registrations

If you performed batch extension creation or if you created a single extension, it is important that you verify a few of the properties of the extension you will use for TopView.

Click the Edit button to the right of the extension number



- Permission: Internal
- AuthID: empty (if empty, the AuthID is the same as the extension number)
- SIP/IAX password: abc5 or your chosen password
- Concurrent registrations: The default is 1 which should (technically) mean that only a single VOIP client can connect using this extension. We've tested this setting and it only appears to affect incoming calls (not supported in TopView VOIP), not outgoing calls from TopView. We recommend setting concurrent registrations to 5 to avoid any potential issues.

General

* Extension:	<input type="text" value="1001"/>	CallerID Number:	<input type="text" value="1001"/>
* Permission:	<input type="text" value="Internal"/>	* SIP/IAX Password:	<input type="text" value="abc5"/>
AuthID:	<input type="text"/>	Voicemail:	<input type="text" value="Local Voicemail"/>
* Voicemail Password:	<input type="text" value="*****"/>	Skip Voicemail Password	<input type="checkbox"/>
Send Voicemail to Email:	<input type="text" value="Default"/>	Verification:	
Enable Keep-alive:	<input type="checkbox"/>	Keep Voicemail after	<input type="text" value="Default"/>
Disable This Extension:	<input type="checkbox"/>	Emailing:	
Emergency Calls CID:	<input type="text"/>	* Keep-alive Frequency:	<input type="text" value="60"/>
		Enable SCA:	<input type="checkbox"/>

User Settings

First Name:	<input type="text"/>	Last Name:	<input type="text"/>
Email Address:	<input type="text"/>	* User Password:	<input type="text" value="*****"/>
* Language:	<input type="text" value="Default"/>	* Concurrent Registrations:	<input type="text" value="5"/>
Mobile Phone Number:	<input type="text"/>		

Analog Trunks

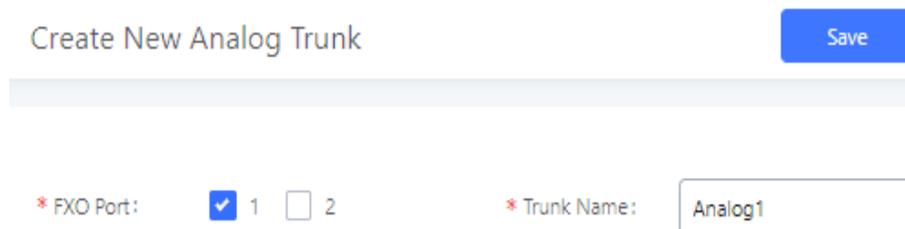
- [Extension/Trunk] from the left menu
- Then select [Analog Trunks]

If you would like to make outgoing calls over an analog line you must first create an analog trunk. The analog trunk is a mapping of a name (the analog trunk name) to an analog phone line/lines plugged into the rear of the UCM.

This UCM model allows up to 2 analog phone lines. Before you can create an analog trunk you need to plug an analog phone line into one of the FXO ports (FXO1 or FXO2) on the rear of the UCM.

*** Be sure to use an FXO port and not an FXS port for the analog phone line**

- Plug the analog phone line into port FXO1 on the rear of the UCM
- [Add+] (or for older firmware [Create New Analog Trunk])
 - FXO port: 1 (if you plugged the analog line into FXO1)
 - Trunk name: Analog1
 - FXO Dial Delay(ms): 1000
 - Some analog lines require a delay between “off hook” and dialing of the first number. The default of 0 ms may not work consistently for calls so we recommend entering a value of 1000 ms (=1 second).
 - Click [Save]
 - Click [Apply changes] at the top of the screen to apply these changes



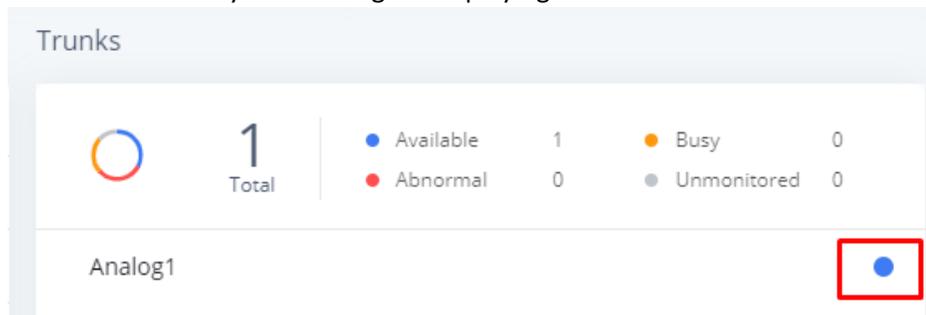
The screenshot shows a web interface for creating a new analog trunk. At the top, the title "Create New Analog Trunk" is displayed in a light blue header bar, with a blue "Save" button to its right. Below the header, there is a light blue horizontal separator line. The main form area contains two fields: "* FXO Port:" with radio buttons for "1" (checked) and "2", and "* Trunk Name:" with a text input field containing "Analog1".

Verify that the analog line is live and can make calls

TopView customers periodically configure the Grandstream with an analog line that is not live or cannot be used by the Grandstream device. This may be due to actual line issues or line filter devices like DSL line filters.

Verify that the Grandstream device sees a live analog line

- [System Status] from the left menu then [Dashboard]
- Under Trunks verify that Analog1 is displaying a blue dot



Test dialing a number over the analog line

Even if the line is live you should verify that you can make calls over the line. This step requires an analog phone.

- Unplug the analog line from the FXO1 port
- Plug the analog line into the analog phone
- Pick up the analog phone (off hook)
 - Verify that you hear a dial tone
 - Verify that you can make calls
- Plug the analog line back into the FXO1 port
- Check the Dashboard screen again and verify that the Analog1 trunk displays a blue dot

Outbound Routes

- [Extension/Trunk] from left menu
- Then select [Outbound Routes]

When a VOIP client (TopView, IP phone, or soft phone) makes a call through the SIP Server using a known extension phone number, that extension will ring. But how can you tell the SIP Server to send the call to the analog line? This is handled through outbound routes.

Routing is handled through pattern matching of the dialed number. The rules for our example are as follows:

- If the called phone number starts with 5 we want the call to go through the analog line
- Once we get to the analog line we are on our internal phone system (PBX). From here I can dial an extension on our internal phone system or 9 to get to an outside line

This example will create an outbound route to send any VOIP call (from TopView) to the analog line (the Analog Trunk named Analog1 created in the previous step) if the dialed number starts with a 5

- Click [Add]
- Calling rule name: SendToAnalog1
- Pattern: _5X. (** Note that the pattern ends with a period! **)
 - _ All patterns start with underscore
 - 5 the called phone number starts with 5
 - X Any digit from 0-9
 - . period = match 1 or more characters (match the X one or more times)
 - Examples:
 - Dial 5102 to call 102 on the analog line
 - Dial 592223333 to call 9 222-3333 on the analog line
- Privilege level: Internal (all extensions can use this rule).
 - Each extension that we created (1001-1010) has a default permission level of Internal. If you want to change the privilege level of this route you need to change the privilege level of the extensions – otherwise they will not be allowed to use the analog line. Go to the extensions and edit an extension to see the permission level.
- Main trunk:
 - Trunk: Analog1 (the analog trunk created in the previous step)
 - Strip: 1 (strip 1 digit from the outgoing number: prevents dialing “5” as the first digit when calling on the analog line)
- Click [Save]
- Click [Apply changes] at the top of the screen to apply these changes

Note: it is possible to route all calls to the analog trunk without requiring a prefix. See “Routing all calls to the analog line”.

* Calling Rule Name:

* Pattern:

PIN Groups:

Password:

Disable This Route:

Privilege Level:

Warning: Setting privilege level at "Internal" has potential security risks.

PIN Groups with Privilege Level:

Enable Filter on Source Caller ID

Enable Filter on Source

Caller ID:

Outbound Route CID:

Call Duration Limit

Call Duration Limit:

Main Trunk

* Trunk:

Strip:

Connect a soft phone to the UCM SIP Server (optional)

To verify that the SIP Server is accessible you can install a desktop or smartphone SIP client (soft phone). You can also use this soft phone to receive test calls from TopView without requiring an analog phone line.

For this example we will use the [3CX phone 6 for Windows](#).

When you first launch the 3CX phone, click the [Create Profile] button then click [NEW] to create a new account. If you have previously created an account you can click on the account name (upper right corner of phone) to edit the settings.

- Account name: UCM (this can be any name)
- Caller ID: enter any name
- Extension: 1002 (this is one of the configured extensions in the UCM)
- ID: 1002
- Password: abc5 (the password for extension 1002 as configured in the UCM)
- IP of the PBX/SIP Server: Select "I am in the office – local IP" and enter 172.16.1.20 (the IP address of the UCM)
- Click [OK] to close the account settings screen.
- On the Accounts screen, make sure that the UCM account is selected (check box before account) and click [OK]
- If the 3CX phone successfully connects to the UCM SIP Service you should see the account name in the upper right corner of the phone and "On Hook" as the status



- Close the 3CX phone.
Note: the phone will remain running until you close it from the system tray.
Right-click the 3CX icon in the system tray and choose [Exit]

Attach an analog phone to the UCM (optional)

You can attach an analog phone (traditional land-line phone) to the UCM using an FXS port. You can assign an extension to the analog phone and make/receive calls using this analog phone just like a soft phone. You can also make test calls from TopView to the analog phone.

- [Extension/Trunk] from the left menu
- Then select [Extensions]

Click [Add]

- Select Extension Type: FXS Extension

This example will connect an analog phone to the FXS1 port in the rear of the UCM and assign the extension 1060 to the analog phone.

- Extension: 1060
- Analog Station: FXS1 (the port in the rear of the UCM where this phone is plugged in)
- Permission: Internal
- Click [Save]
- Click [Apply changes] at the top of the screen to apply these changes
- Pick up the analog phone to verify that you can hear a dial tone
- If you installed the soft phone to extension 1002 you should be able to call the soft phone by dialing 1002. If you closed the 3CX phone in the previous step you will need to restart it.

General

* Extension:	<input type="text" value="1060"/>	Analog Station:	<input type="text" value="FXS 1"/>
CallerID Number:	<input type="text"/>	* Permission:	<input type="text" value="Internal"/>
Voicemail:	<input type="text" value="Local Voicemail"/>	* Voicemail Password:	<input type="password" value="....."/>
Skip Voicemail Password <input type="checkbox"/>		Send Voicemail to Email:	<input type="text" value="Default"/>
Verification:		Keep Voicemail after:	<input type="text" value="Default"/>
		Emailing:	
Disable This Extension: <input type="checkbox"/>		Emergency Calls CID:	<input type="text"/>

User Settings

First Name:	<input type="text" value="Analog phone"/>	Last Name:	<input type="text"/>
Email Address:	<input type="text"/>	* User Password:	<input type="password" value="*****"/>
* Language:	<input type="text" value="Default"/>	Mobile Phone Number:	<input type="text"/>

Make test calls from TopView

- Start the TopView Configurator
- Open an existing configuration or start a new configuration
- From the left menu select “Voice Notification”
- On the screen check “Enable Voice Notification” and click [Configure]
 - Check “Use VOIP” if not selected
 - SIP Server host/IP: 172.16.1.20 (the IP address of the UCM)
 - Proxy: (leave blank)
 - Display name: TopView (this can be any name)
 - User name: 1005 (this is the extension in the UCM that we want to use for TopView)
 - Authorization name : 1005 (same as user name if extension AuthID is blank)
 - Password: abc5 (the password for extension 1005 as configured in the UCM)
 - Registration required: should be checked
- Make a test call using the [Make test call] button
 - If you installed a soft phone in the previous steps (extension 1002) you can make a test call to 1002 and the soft phone should ring. Answer the phone to hear the message and acknowledge (#).
 - If you installed an analog phone in the previous steps (extension 1050) you can make a test call to 1050 and the phone should ring. Answer the phone to hear the message and acknowledge (#).
 - To make a test call through the analog line, enter the phone number starting with 5. The “5” is the outgoing route for the analog line and will be stripped when making the call over the analog line. All other digits after “5” will be used to make the call on the analog line.

Note: on the TopView call settings screen for Voice Notification you can enter a “dial prefix” to instruct TopView to prefix all calls with a 5. If all calls from TopView will be made to the analog line you can enter the dial prefix and do not need to add the prefix to called phone numbers.

Admin and Technical Notes

Routing all calls to the analog line/trunk

In the example configuration for the outbound route we use a pattern that requires the use of a “5” prefix to all numbers that should be called through the analog line. See “Outbound Routes”.

The entered pattern for the outbound route can be used to route all calls to the analog line without a special prefix.

Example: route all phone numbers to the analog trunk

- Pattern: _X. (1 or more digits)
- Strip: 0 (do not strip any characters from the entered number)