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EE/CS 52 VoIP
TA: Jennifer Lin
1/15/2017

VoIP Functional Specification

Description:

The VoIP phone is a standard VoIP phone that communicates over SIP+RTP on the standard port ranges (SIP:5060-5061, RTP:16384-32767). When the phone is powered on, it attempts to acquire an IP, subnet mask, and gateway through DHCP. If this fails, the user is able to configure these settings manually. VLAN tagging on packets is ignored (since there is no second Ethernet port to forward packets through). To make a call, the user dials a valid 11 digit number and then waits for two seconds. To hang up, the user presses the hangup button.

Global Variables:

None

Inputs:

The user input is through a 4x4 array of buttons, a microphone, and a phone hangup button. Through this input, the user is able to dial a number, and talk to the dialed phone.

Outputs:

The phone has three modes of output, a speaker/headphone jack, an lcd display, and an Ethernet jack to the internet (and another VoIP phone).

Speaker/Headphone jack:

The speaker is a 1 watt speaker. If headphones are plugged in, the speaker is disabled, and audio is sent to the headphones.

LCD Display:

The LCD Display is a 128x32 LCD gray display.

Ethernet Jack:

The Ethernet jack is a standard RJ45 jack. The Ethernet data rate is either 10 or 100 Mbps.

User Interface:

The user input is through a 4x4 array of buttons, a microphone, and a phone hangup button.

The button layout is:

Menu	**1**	**2/Up**	**3**
Volume Up	**4/Left**	**5/Enter**	**6/Right**
Volume Down	**7**	**8/Down**	**9**

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Reset	*****	**0**	**#**

Button Descriptions:

Menu – toggles the settings menu. See “Settings Menu” below.

Volume Up/Down – adjusts the volume level.

Reset – resets the phone.

0-9 – number keys for dialing a phone number.

Up/Left/Right/Down – used in the settings menu for navigating the menu.

Enter – used in the settings menu to confirm a setting.

***/#** - characters used for dialing.

Menu:

By clicking the menu button, the user can bring up a one-dimensional array of options they can scroll through to set the phone IP, subnet mask, and gateway, or save/recall phone numbers. The order of the menu is:

1. Set IP
2. Set Subnet Mask
3. Set Gateway
4. Save Number
5. Recall number

The menu is navigated with the 2-4-5-6-8 keys, which are Up-Left-Enter-Right-Down, respectively. To exit the menu, the menu button is pressed again. The save number and recall number menus allow the user to save the current number into locations 0-9, or recall the number from locations 0-9.

In addition to this 4x4 array of buttons, there is a separate button for hanging up the phone. On a normal phone, this would be the button that detects whether the phone is in its holder.

The microphone is connected to the board through a standard 3.5mm headphone jack.

Output:

The phone has three modes of output, a speaker/headphone jack, an lcd display, and an Ethernet jack to the internet (and another VoIP phone).

The speaker output can be either a speaker built onto the board, or a 3.5mm headphone jack. If headphones are plugged into the board, audio will be played through the headphone jack, else, audio will be played through an onboard speaker.

The LCD display will by default display the current number being dialed, or nothing if no number is being dialed.

Error Handling:

If an invalid number is dialed, or the dialing can't be completed, the phone plays an error tone.

Algorithms:

Only one gateway is supported, so the destination IP is compared with the subnet mask to determine whether the packet should be sent to the default gateway at the MAC layer, or directly to the destination at the MAC layer.

Data Structures:

Voice data is sent in RTP packets

Limitations:

Only one internet gateway is supported.

Known Bugs:

None known

Special Notes:

None

Versioning:

Version	Owner	Comment	Date
1.A	Will Werst	Initial Version	1/15/2017