Voice and Text to Speech on the BBB and RPi2/3

This document explains several ways you can achieve text to speech in the hamvoip versions of Allstar.

Text to Speech (TTS)

Versions BBB 1.2 and RPi2 1.0 had a TTS script that directly called the Google TTS engine. About mid-year 2015 Google decided to limit the use of this service based on the number of times you accessed it. There was no option to pay for it instead it was just cutoff after excessive use. So many who are still using these versions of the code will not, or eventually will not, have access to TTS services for ID's, Weather, or other uses.

Version 2.0 for the BBB and RPi2/3 will reintroduced TTS using the ***voicerss.org*** service. Although not as good as Google it is better than any current Linux speech. This service can be used to generate ID's or for other text to speech uses such as repeater messages, etc.

To download the new version in **pre-2.0 releases** and overwrite the old non working Google code do the following login to your Allstar server and at the Linux prompt do the following:

cd /root
wget https://hamvoip.org/downloads/hamvoip-tts-scripts-0.2-6.pkg.tar.xz
pacman -U --force hamvoip-tts-scripts-0.2-6.pkg.tar.xz

It should install and you can now proceed with the setup below -

Using the voicerss system

The ***voicerss.org*** TTS is free for most use but you must sign up at the site and get an API code. Once you get this API code you need to insert it into the ***/usr/local/etc/tts.conf*** file. The file looks like this:

```
# TTS Audio Configuration file.

# This file contains the tts_audio configruation parameters.
# If this is not setup correctly, or they key is incorrect, the
# tts_audio.sh script will not function
#
# For getting a key, you will need to register at:
#     www.voicerss.org
#
# Once registered, you will receive a API key which needs to be
# added below in the "tts_key=" entry.
#
# You can also change the speech rate and language. For more information,
# on these settings go to www.voicerss.org.

tts_key=
```

 # key for voicerss.org url, this is manditory

tts_r=-1

 # tts_r is speech rate (speed). Allows values:
 # from -10 (slowest speed) up to 10
 # (fastest speed).

tts_hl=en-us

 # tts_hl is the language setting
 # See voicerss.org documentation for further
 # details of what languages are supported.

tts_f=44khz_16bit_mono

 # The speech audio formats. Allows values:
 # see Audio Formats on voicrss.org

Insert the key you were assigned at the voicerss.org site in the tts_key= line. Only add the key, no other quotes or characters.

Example - **tts_key=bvdsd6adsd3sds7898gmhukuk9gr665**

This is NOT a real key but shows how a key should be entered. It is best to cut and paste the key if possible for accuracy.

Once the key is inserted save the ***tts.conf*** file.

You can then start using the ***tts_audio.sh*** script which is in the global Linux path. For backup purposes it is recommended you create and save user files in the ***/etc/asterisk/local*** directory. Create a text file and save it. For an example we will call it ***myid.txt*** and the text file will have the following contents but of course using your information and in the way you want to say it. We will use ***nano*** to create the file.

cd /etc/asterisk/local

nano myid.txt

W A 3 D S P, Allstar, Richboro, Pennsylvania, P L 88.5

Control X and save

Note the when entering single letters you put a space. If WA3DSP was entered it would probably try to pronounce it as one word and fail miserably.

If all went well, the audio file ***myid.ul*** will be created in the same directory. If there is a failure, see the possible errors and causes below.

The created audio file can be directly played by Allstar using the localplay or playback commands. Note that playback plays the file globally to every node you are connected to!! This method would generally not be used unless you wanted to reach a broad audience.

Here is an example playing the created file to node 1998 with ***myid.ul*** in ***/etc/asterisk/local***

Enter the Asterisk client at the Linux prompt -

asterisk -rvvv

At the CLI prompt enter the command to play the file.

CLI> rpt localplay 1998 /etc/asterisk/local/myid

This is an excellent way to test play the file on your local radio.

You could also play the file directly from Linux or in a script, cron job etc. like this:

asterisk -rx "rpt localplay 1998 /etc/asterisk/local/myid"

Note that you never use the file extent (.ul etc.) in the play statements. Replace **localplay** with **playback** for global play.

We used the name ***myid.txt*** which became ***myid.ul*** here but the Allstar default voice ID in the ***rpt.conf*** file looks for the file name ***/etc/asterisk/local/node-id*** as the voice ID file. So you either need to change the file name to ***node-id*** or the or the name that is called in ***rpt.conf*** to the name of the file you want to be your ID.

<u>Possible TTS Errors</u>

At times the ***voicerss.org*** site may not be available and a file will not be created. If this is the case try again later. Also make sure you are registered at the site and the proper API has been inserted in the ***tts.config*** file.

<u>Tips on creating TTS files</u>

Most of the time the audio rendition of the text file sounds quite good but in some cases you might need to experiment with spelling and word placement. Here are a few tips.

When entering a call space the characters. The tts processor will try to sound out WA3DSP if entered in that manner which would sound like gibberish. Instead type W A 3 D S P (spaces in between) which will say each character separately.

Typing - P L 131.8 will say 'PL one hundred thirty one dot eight' To say it the way we are use to hearing it type - P L 1 31.8 or P L one thirty one dot eight

Commas and periods leave extra space and often strategically placed make things sound better.

Sometime misspelling a word can make it sound more like you want. This is especially true for town names. One online TTS program would not say 'A' correctly so spelling it 'aye' seemed to work.

Initial Setup of Voice ID

During the initial setup of Allstar the TTS audio as described above will not be configured so the options will either be the Asterisk Allison voice of your call letters or just a CWID. Select the option which suits you. You can later come back and create a custom TTS ID of your liking once you have configured the TTS script as described above.

Combining Allison voice files

Another way to create voice files is to combine Allison phrases to say what you want. This is the way it is done in the initial setup only just your call is defined at that time. The ***combine.sh*** script in ***/etc/asterisk/local*** is an example of this. ***combine.sh*** creates an ID by combining the letters and numbers of your call and any other sound from the Allison sound library. The only requirement is that all the phrases be of the same type – gsm, ulaw, etc

So if you want to say WA3XYZ PL 88.5 the script would look like this:

cat /var/lib/asterisk/sounds/letters/w.gsm /var/lib/asterisk/sounds/letters/a.gsm /var/lib/asterisk/sounds/digits/3.gsm /var/lib/asterisk/sounds/letters/x.gsm /var/lib/asterisk/sounds/letters/y.gsm /var/lib/asterisk/sounds/letters/z.gsm /var/lib/asterisk/sounds/letters/p.gsm /var/lib/asterisk/sounds/letters/l.gsm /var/lib/asterisk/sounds/digits/80.gsm /var/lib/asterisk/sounds/digits/8.gsm /var/lib/asterisk/sounds/letters/dot.gsm /var/lib/asterisk/sounds/digits/5.gsm > /etc/asterisk/local/**node-id.gsm**

Note that this is one complete line of text and of course you could shorten it up considerably using variables to represent file paths.

Executing this script produces the audio file ***node-id.gsm*** which can be played for the voice ID in ***rpt.conf***.

The asterisk sounds or phrases are located in the ***/var/lib/asterisk/sounds*** directory. There you will find many words and phrases. One example is the word “repeater” which could be added to your ID.

Other Voice Methods

There are numerous voice packages such as Festival, Espeak, and commercial software such as that available from Cepstral and AT&T. Many have been investigated and none of the free packages come anywhere near the quality of online TTS products. There are commercial packages that sound quite good by carry a significant price tag.

You have to define your voice requirements. Are they dynamic or static? Meaning is it changing text that you need to convert on the fly such as a weather forecast or is it predefined text such as an ID or standard message that you can record ahead of time and reuse.

Here are some methods to record static messages.

There are a myriad of text to speech readers for Windows which can be used to create speech. Many allow saving to a file but if not you can always save what it says with a program like audacity and then process the file for use with Allstar. Here is a link to some of the popular Windows TTS readers. Note that some recent versions of Windows may have TTS built-in as an option.

http://alternativeto.net/software/espeak/

There are also many online TTS web pages that allow you to enter text and play it back. Here is one we use for the pre-recorded clips for the weather scripts. It allows you to download the file which you would then process for Asterisk Allstar.

http://www.fromtexttospeech.com/

You always have the option of recording your own or someone you pick real voice recorded with a microphone and converted to use with Asterisk Allstar. Here are a few links that might help you with this method.

http://www.itp-redial.com/class/weekly-notes/week2-notes/recording-audio-for-asterisk

http://blogs.digium.com/2011/04/19/asterisk-sound-files-101/

If you would like to put a face on Allison the woman behind Asterisk sounds here she is -

https://www.digium.com/products/ivr/allison-smith

One of the most important things when recording a file is to use a quality microphone in a quiet environment. Also use at least 22Khz 16 bit mono recording. You could use higher if desired. It is always better to start out at a higher rate and re-sample down. Because Asterisk can only use an 8Khz sample rate it must be converted before use. To do this you could use an online converter or sox on your BBB or RPi2.

Online converter -

http://my.digium.com/en/products/ivr/audio-converter/

To use **sox** to convert the higher sample rate wav file to a ul type at 8khz playable on Asterisk Allstar:

sox -V audio.wav -r 8000 -c 1 -t ul audio.ul