Fax For Asterisk™

Administrator Manual

601-00012 Rev. D3

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Digium, Inc. has made every effort to ensure that the instructions contained in this document are adequate and error free. The manufacturer will, if necessary, explain issues which may not be covered by this documentation. The manufacturer's liability for any errors in the documents is limited to the correction of errors and the aforementioned advisory services.

This document has been prepared for use by professional and properly trained personnel, and the customer assumes full responsibility when using it.

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Chapter 1: Overview

Digium's Fax For Asterisk™ is a commercial facsimile (fax) termination and origination solution designed to enhance the capabilities of Open Source and commercial Asterisk, as well as Switchvox. Fax For Asterisk bundles a suite of user-friendly Asterisk applications and fax modem software. Fax For Asterisk provides low speed (14,400 bps) PSTN faxing via DAHDI-compatible telephony boards as well as VoIP faxing to T.38-compatible SIP endpoints and service providers. Licensed on a per-channel basis, Digium's Fax For Asterisk provides a complete, cost-effective, commercial fax solution for Asterisk users.

Fax For Asterisk provides two components: res_fax and res_fax_digium. The res_fax Asterisk resource module adds fax termination and origination functionality in Asterisk. It provides the FAXOPT Asterisk dialplan function and the SendFAX and ReceiveFAX dialplan applications to enable the user to build highly-customizable fax solutions. The res_fax_digium Asterisk resource module provides core fax processing functionality in the form of T.38 support and several supported fax modems – V.21, V.27ter, V.29, and V.17 – which achieve speeds up to 14,400 bps.

Fax For Asterisk provides the functionality to send and receive faxes to and from TDM and IP channels – TDM channels are established across Digium telephony boards, and IP channels using T.38 encapsulation. Faxes transmitted and received by Fax For Asterisk begin and end as TIFF image files. TIFF files may be readily converted into or from other formats using standard Linux command-line utilities.

Digium's customers of Fax For Asterisk may purchase license keys coded for a specific number of channels. Each licensed channel allows Fax For Asterisk to initiate one modem session or process one fax session. As customers need to expand their fax capacity, they may purchase additional license keys to register on their existing Asterisk system. The aggregate number of channels across all registered license keys will be made available to Asterisk.

Each open source or commercial Asterisk system is eligible to receive from Digium, a single channel of Fax For Asterisk, called Free Fax For Asterisk, for no cost. Free Fax For Asterisk is provided under license as-is, without technical support, and is available to all Asterisk users as a free, zero cost purchase from the Digium webstore. Only one channel of Free Fax For Asterisk may be used with an installation of Asterisk. If you require multiple channels of Fax capability or if you require Digium's technical support, you may purchase channels of Fax For Asterisk from *http://www.digium.com* .

1.1 What is Asterisk®?

Asterisk is the world's leading open source telephony engine and tool kit. Offering flexibility unheard of in the world of proprietary communications, Asterisk empowers developers and integrators to create advanced communication solutions...for free. Asterisk is released as open source under the GNU General Public License (GPL), and it is available for download free of charge. Asterisk is the most popular open source telephony software available, with the Asterisk Community being the top influencer in VoIP.

1.2 Asterisk as a Phone Switch (PBX)

Asterisk can be configured as the core of an IP or hybrid PBX, switching calls, managing routes, enabling features, and connecting callers with the outside world over IP, analog (POTS), and digital (T1/E1/J1/BRI) connections. Asterisk runs on a wide variety of operating systems including Linux, Mac OS X, OpenBSD, FreeBSD, and Sun Solaris. It provides all of the features you would expect from a PBX including many advanced features that are often associated with high end (and high cost) proprietary PBXs. Asterisk's architecture is designed for maximum flexibility and supports Voice over IP in many protocols, and can interoperate with almost all standards-based telephony equipment using relatively inexpensive hardware.

1.3 Asterisk as a Gateway

It can also be built out as the heart of a media gateway, bridging the legacy PSTN to the expanding world of IP telephony. Asterisk's modular architecture allows it to convert between a wide range of communications protocols and media codecs.

1.4 Asterisk as a Feature/Media Server

Need an IVR? Asterisk's got you covered. How about a conference bridge? Yep. It's in there. What about an automated attendant? Asterisk does that too. How about a replacement for your aging legacy voicemail system? Can do. Unified messaging? No problem. Need a telephony interface for your web site? Okay.

1.5 Asterisk in the Call Center

Asterisk has been adopted by call centers around the world based on its flexibility. Call center and contact center developers have built complete ACD systems based on Asterisk. Asterisk has also added new life to existing call center solutions by adding remote IP agent capabilities, advanced skills-based routing, predictive and bulk dialing, and more.

1.6 Asterisk in the Network

Internet Telephony Service Providers (ITSPs), Competitive Local Exchange Carriers (CLECs) and even first-tier incumbents have discovered the power of open source communications with Asterisk. Feature servers, hosted services clusters, voicemail systems, and pre-paid calling solutions, all based on Asterisk have helped reduce costs and enabled flexibility.

1.7 Asterisk Everywhere

Asterisk has become the basis for thousands of communications solutions. If you need to communicate, Asterisk is your answer. For more information on Asterisk, visit *http://www.asterisk.org* or *http://www.digium.com*.

Chapter 2: Installation

This chapter will guide you through the necessary steps to install Digium's Fax For Asterisk.

- **Install Notes:**
    - T.38 Termination is only available in the res_fax and res_fax_digium modules for the Open Source Asterisk 1.6 (or later) releases. This is because the Open Source Asterisk 1.4 releases do not support T.38 Termination.
    - If you will be using an Open Source Asterisk 1.4 release, Digium recommends using Open Source Asterisk 1.4.22 or newer. Versions prior to 1.4.22 have not been tested.
    - If you will be faxing over TDM, Digium recommends using DAHDI 2.1.0.3 or newer. Versions prior to 2.1.0.3 have not been tested.
    - Digium recommends a minimum version for the various offerings of Asterisk. The recommendations are provided in the table shown below. Versions prior to those recommended have not been tested.

Asterisk	Recommended Minimum Version
Open Source Asterisk branch 1.4	1.4.22
Open Source Asterisk branch 1.6.0	1.6.0.14
Open Source Asterisk branch 1.6.1	1.6.1.5
Open Source Asterisk branch 1.6.2	1.6.2.0

2.1 Installation Overview

Once you have your Fax For Asterisk license key, there are a few tasks to perform in order to install Fax For Asterisk.

1. Download and execute the *register* utility to generate a valid license.
2. Download and execute the *benchfax* utility to determine the optimum build.
3. Use the *Fax Selector* web utility to determine your required components.
4. Download and install the *res_fax* binary that is built for your platform.
5. Download and install the *res_fax_digium* binary that is built for your platform.

The register utility may be downloaded from:
http://downloads.digium.com/pub/register/

The benchfax utility may be downloaded from:
http://downloads.digium.com/pub/telephony/fax/benchfax/

The Fax Selector web utility may be accessed from:
http://www.digium.com/en/docs/FAX/faa-download.php

The res_fax binary may be downloaded from:
http://downloads.digium.com/pub/telephony/fax/res_fax/

The res_fax_digium binary may be downloaded from:
http://downloads.digium.com/pub/telephony/fax/res_fax_digium/

> **Note:** Supported software builds are provided for 32-bit and 64-bit x86 platforms, and are optimized for a variety of processor types. Choose the directory that closest matches your Asterisk version and processor type. Each of these directories contains TAR files which include the fax modules for each type of supported processor.

2.2 Register Fax For Asterisk

Registration of the Fax For Asterisk license key will be done using the Digium register utility in the same way as with other modules like Cepstral, HPEC, and G.729. The registration utility will prompt you for your Fax For Asterisk license key. An example for 32-bit Linux has been provided below. Be sure to log in as the user “root” before executing similar commands.

```
# cd /root
```

```
# wget http://downloads.digium.com/pub/register/x86-32/register
# chmod 500 /root/register
# /root/register
```

Notes:

- Internet access is required from your Asterisk server in order to register your Fax For Asterisk key for licensed use. Outgoing network traffic on TCP port 443 (SSL) must be allowed in order for the register utility to successfully communicate with Digium's license server and complete the registration process. You must have at least one Ethernet device in your Asterisk server in order for the registration process to complete successfully.

- Multiple Fax For Asterisk keys may be registered on the same Asterisk server. This will allow you to increase the total number of available Fax For Asterisk channels on your Asterisk server. New Fax For Asterisk keys may be registered to your Asterisk server using the same instructions provided above. There will be an additional Fax For Asterisk license file generated in the /var/lib/asterisk/licenses directory for each Fax For Asterisk key that is successfully registered to your Asterisk server. It is extremely important that you follow the instructions provided in section *2.9* whenever a new Fax For Asterisk key is successfully registered to your Asterisk server.

- A Fax For Asterisk key must be re-registered if any of the Ethernet devices in your Asterisk server are changed, added, or removed. The unique Fax For Asterisk license file which is located in your /var/lib/asterisk/licenses directory is tied to the MAC address of all the Ethernet devices installed in your system. A Fax For Asterisk key can only be re-registered once without authorization from Digium. Digium must be contacted by phone in order to request authorization to have your Fax For Asterisk key incremented. Digium reserves the right to deny authorization for having a Fax For Asterisk key incremented. Please note that Digium will not provide assistance with Free Fax For Asterisk keys; support is only provided for paid Fax For Asterisk customers.

2.3 Determine Optimum Build

There are various optimized versions of the fax module available for different CPU types in x86-32 and x86-64 architectures. To determine which build of the module performs best on your system, the *benchfax* utility will run a series of tests, and report which fax module will maximize performance on your system. An example has been provided below.

```
# wget http://downloads.digium.com/pub/telephony/fax/benchfax/\
  x86-32/benchfax-1.1.0-x86_32 -O benchfax
# chmod 500 /root/benchfax
# /root/benchfax
```

2.4 Determine Required Components

Depending upon your version of Asterisk and processor architecture, different components are required for the use of Fax For Asterisk. Digium provides a *Fax Selector* web utility in order to assist with choosing the correct components. The *Fax Selector* web utility should be viewed using a standard web browser and may be accessed via the following URL:

http://www.digium.com/en/docs/FAX/faa-download.php

The files that the *Fax Selector* web utility informs you are required for your platform are the ones that you should use in place of the following installation examples.

2.5 Install res_fax

There are different versions of *res_fax* for various Asterisk releases; there is a single version for Asterisk 1.4.22 and above, and there are versions for Asterisk 1.6.x point releases (1.6.0, 1.6.1, etc.). Take note that these modules are **not** loadable in prior releases of Asterisk, only the specific version they are designed to be used with. Please be sure that you download the correct version of *res_fax* for your Asterisk version as recommended by the *Fax Selector* web utility. Be aware that the *Fax Selector* web utility may indicate that your version of Asterisk does not require that you download and install the *res_fax* module.

If the *Fax Selector* web utility did not indicate that a *res_fax* download is required for your system, please skip to section *2.6*.

There are frequently updated builds of *res_fax* posted, and each build has a *version number*. This version number is part of the filename, and is also included in the copyright/license message that is displayed when the module is loaded into Asterisk. In this document, build number *1.2.1* has been used as an example, but when you read this document the current build number may be different (higher).

The *res_fax* module must be extracted and placed in Asterisk's modules directory (default is /usr/lib/asterisk/modules). An example has been provided below.

```
# wget http://downloads.digium.com/pub/telephony/fax/res_fax/\
  asterisk-1.6.0/x86-32/res_fax-1.6.0_1.2.1-x86_32.tar.gz
# tar xzvf res_fax-1.6.0_1.2.1-x86_32.tar.gz
```

```
# cp /root/res_fax-1.6.0_1.2.1-x86_32/res_fax.so \
  /usr/lib/asterisk/modules
```

2.6 Install res_fax_digium

There are different versions of *res_fax_digium* for various Asterisk releases; there is a single version for Asterisk 1.4.22 and above, and there are versions for Asterisk 1.6.x point releases (1.6.0, 1.6.1, etc.). Take note that these modules are **not** loadable in prior releases of Asterisk, only the specific version they are designed to be used with. Please be sure that you download the correct version of *res_fax_digium* for your Asterisk version as recommended by the *Fax Selector* web utility.

There are frequently updated builds of *res_fax_digium* posted, and each build has a *version number*. This version number is part of the filename, and is also included in the copyright/license message that is displayed when the module is loaded into Asterisk. In this document, build number *1.2.1* has been used as an example, but when you read this document the current build number may be different (higher).

The *res_fax_digium* module must be extracted and placed in Asterisk's modules directory (default is /usr/lib/asterisk/modules). An example has been provided below.

```
# wget http://downloads.digium.com/pub/telephony/fax/res_fax_digium/\
  asterisk-1.6.0/x86-32/res_fax_digium-1.6.0_1.2.1-pentium4m.tar.gz
# tar xzvf res_fax_digium-1.6.0_1.2.1-pentium4m.tar.gz
# cp /root/res_fax_digium-1.6.0_1.2.1-pentium4m/res_fax_digium.so \
  /usr/lib/asterisk/modules
```

2.7 Load Fax For Asterisk Modules

The res_fax and res_fax_digium Asterisk resource modules must be loaded in Asterisk in order to use the Fax For Asterisk channels. An example is provided below.

```
*CLI> module load res_fax.so
*CLI> module load res_fax_digium.so
```

If you already have res_fax_digium.so loaded and have registered a new license key to increase the number of Fax For Asterisk channels, simply reload the module using the following command.

```
*CLI> module reload res_fax_digium.so
```

Reloading this module will only be successful if no fax sessions are in progress. If there are active fax sessions, you will either have to wait until they have completed to manually reload the module, or schedule Asterisk to restart once there are no active calls by executing the following command.

```
# asterisk -rx “restart when convenient”
```

2.8 Verify Installation

Verify that the number of Fax For Asterisk channels available to Asterisk matches the number of Fax For Asterisk channels that you purchased. This can be verified by issuing "fax show stats" in the Asterisk CLI. Take into consideration any previous Fax For Asterisk channels that you may have already had registered to your Asterisk server before verifying this number. An example is provided below.

```
# asterisk -rvvv

*CLI> fax show stats

     Fax Statistics:
     ---------------

     Current Sessions     : 0
     Transmit Attempts    : 0
     Receive Attempts     : 0
     Completed Faxes      : 0
     Failed Faxes         : 0

     Digium T.38
     Licensed Channels    : 200
     Max Concurrent       : 0
     Success              : 0
     Canceled             : 0
     No Fax               : 0
     Partial              : 0
     Negotiation Failed   : 0
     Train Failure        : 0
     Protocol Error       : 0
     IO Partial           : 0
     IO Fail              : 0

     Digium G.711
     Licensed Channels    : 200
     Max Concurrent       : 0
     Success              : 0
     Switched to T.38     : 0
     Canceled             : 0
     No Fax               : 0
     Partial              : 0
```

```
Negotiation Failed   : 0
Train Failure        : 0
Protocol Error       : 0
IO Partial           : 0
IO Fail              : 0
```

2.9 Backup License File

It is extremely important that you backup all of the files located in the /var/lib/asterisk/licenses directory. This directory contains the Host-ID specific license files for your system. These license files are tied to the MAC address of all the Ethernet devices installed in your system. Creating a backup of this directory will allow you to restore your Fax For Asterisk licenses in case you need to reinstall your operating system.

> **Note:** A Fax For Asterisk key must be re-registered if any of the Ethernet devices in your Asterisk server are changed, added, or removed. A Fax For Asterisk key can only be re-registered once without authorization from Digium. Digium must be contacted by phone in order to request authorization to have your Fax For Asterisk key incremented. Digium reserves the right to deny authorization for having a Fax For Asterisk key incremented. Please note that Digium will not provide assistance with Free Fax For Asterisk keys; support is only provided for paid Fax For Asterisk customers.

Chapter 3: Configuration

Digium's Fax For Asterisk has a variety of configuration options. This chapter provides an explanation of the configuration options which are available.

3.1 Application Interfaces

The FAXOPT dialplan function and the SendFAX and ReceiveFAX Asterisk applications provide fax interfaces to the Asterisk dialplan. Arguments are used to specify fax transmission details like input/output fax file and the enablement of debug or ECM. The following sections detail the options for the FAXOPT function and the fax applications.

3.1.1 FAXOPT Function

The FAXOPT dialplan function is used to set options to be used by the SendFAX and ReceiveFAX applications, and to read results after a SendFAX or ReceiveFAX application completes. The options available to FAXOPT are listed in the table below.

Option	Type	Description	Values
ecm	RW	Specify the Error Correction Mode (ECM)	yes | no
error	RO	Read fax transmission failure reason	<error>
filename	RO	Read the filename used during the fax transmission. Limited to displaying a single filename.	<filename>
filenames	RO	Read the filenames used during the fax transmission. The output is comma separated.	<filenames>
headerinfo	RW	Specify the fax header	<string>
localstationid	RW	Specify the local station identification	<string>
maxrate	RW	Specify the maximum transfer rate to be used during the fax transmission rate negotiation	2400 | 4800 | 7200 | 9600 | 12200 | 14400
minrate	RW	Specify the minimum transfer rate to be used during the fax transmission rate negotiation	2400 | 4800 | 7200 | 9600 | 12200 | 14400
modem	RW	Specify the modem(s) capabilities for a session. Specification of a single modem will force that modem type. Specification is a comma-separated list of one or more of the possible values	V17 | V27 | V29 | V34
pages	RO	Read the number of pages transferred during the fax transmission	<int>
rate	RO	Read the negotiated fax transmission rate	2400 | 4800 | 7200 | 9600 | 12200 | 14400
remotestationid	RO	Read the remote station identification	<string>
resolution	RO	Read the image negotiation	<image resolution>
status	RO	Read the result status of the fax transmission	SUCCESS | FAILED
statusstr	RO	Read a verbose result status string of the fax transmission	<statusstr>

FAXOPT options of RW (Read/Write) types are written before the fax transmission and are typically either readable after the fax transmission has completed or following a FAXOPT write operation. FAXOPT options of RO (Read Only) types can only be read following the completion of a SendFAX or ReceiveFAX application. FAXOPT options of WO (Write Only) types can only be written prior to the initiation of SendFAX or ReceiveFAX. Below are some descriptions of options that may not be intuitive.

- **ecm** – Error Correction Mode (ECM) enable/disable option. This application argument is used to specify or override the current default configuration setting. The default setting is ECM enabled. See section *3.2* for a list of valid settings.
- **localstationid** – Local station identification. Text string that identifies the sender identification to the remote side of the fax transmission.
- **maxrate** – Maximum transfer rate used during fax rate negotiation. See section *3.2* for a list of valid settings. The default maximum transfer rate is 14400.
- **minrate** – Minimum transfer rate used during fax rate negotiation. See section *3.2* for a list of valid settings. The default minimum transfer rate is 2400.

- **modem** – A comma separated list of one or more of the possible values. The default value is “V17,V27,V29”. This option is intended to replace the deprecated FAX_FORCE_xx and FAX_DISABLE_xx channel variables.

3.1.2 SendFAX Application

The SendFAX application is the default application for sending one or more fax files. The 1.4 version of res_fax provides a *rxtxappnames* configuration option that is intended to ease conversion to res_fax from spandsp-based applications. Anywhere this document refers to *SendFAX* implies the optional use of *TxFAX* for the 1.4 version of res_fax if /etc/asterisk/res_fax.conf includes *rxtxappnames=yes*. The following section describes the SendFAX interface.

```
SendFAX(<filename[&filename2&filename3&...]>[,ad])
      <filename> : Full path to the TIFF image to transmit. If sending
      multiple fax files, append each additional full path using the
      ampersand (&).
      'd' – Enables fax debug reporting. More granular event reporting will
      be observed when 'verbose' logging is enabled in Asterisk. Manager
      sessions will receive manager events for each granular fax session
      event. This is an optional argument. (default: off)
      'f' – Allow fax fallback to audio mode on T.38-capable channels
      'z' – initiate a T.38 reinvite on the channel if the remote end does
      not
      's' – Send progress Manager events (overrides statusevents setting in
      res_fax.conf)
```

The following input channel variables are used by SendFAX for backwards compatibility with previous spandsp-based Asterisk applications. New development and dialplan creators should use the FAXOPT dialplan function. FAXOPT will override channel variables in the slim case that both FAXOPT and a channel variable were used for the same call to SendFAX or ReceiveFAX; *e.g.* FAXOPT(headerinfo) and LOCALHEADERINFO.

- **LOCALSTATIONID** – Text string that identifies the sender identification to the remote side of the fax transmission.
- **LOCALHEADERINFO** – Text string that becomes the fax header sent on each page. If this variable is not set, no header will be used.

The 1.4 version of res_fax supports the following channel variables for backwards compatibility, but it should be noted that these channel variables are deprecated and not supported in 1.6 or newer versions of res_fax.

- **FAX_DISABLE_V17** – Set to '1' to disable V.17.
- **FAX_FORCE_V17** – Set to '1' to force V.17.
- **FAX_FORCE_V27** – Set to '1' to force V.27.
- **FAX_FORCE_V29** – Set to '1' to force V.29.

- **FAX_FORCE_V34** – Set to '1' to force V.34.
- **PHASEESTATUS** – This channel variable will always have the same value as the FAXSTATUS channel variable and is in only intended to ease conversion to res_fax.
- **PHASEESTRING** – This channel variable will always have the same value as the FAXSTATUSSTRING channel variable and is only intended to ease conversion of res_fax.

The following output variables are set by SendFAX when the fax transmission completes.

- **FAXSTATUS** – The fax operation result.
- **FAXERROR** – The reason for a fax failure.
- **FAXSTATUSSTRING** – The fax operation result string.
- **REMOTESTATIONID** – Text string that identifies the remote station.
- **FAXPAGES** – The number of pages transferred during the fax transmission.
- **FAXBITRATE** – The transmission rate used for the fax transmission.
- **FAXRESOLUTION** – The fax image resolution used for the fax transmission.

3.1.3 ReceiveFAX Application

The ReceiveFAX application is the default application for receiving a fax file. The 1.4 version of res_fax provides a *rxtxappnames* configuration option that is intended to ease conversion to res_fax from spandsp-based applications. Anywhere this document refers to *ReceiveFAX* implies the optional use of *RxFAX* for the 1.4 version of res_fax if /etc/asterisk/res_fax.conf includes *rxtxappnames=yes*. The following section describes the ReceiveFAX interface.

```
ReceiveFAX(<filename>[,cd])
      <filename> : Full path to the file to receive, overwrite if file
      already exists
      'd' - Enables fax debug reporting. More granular event reporting will
      be observed when 'verbose' logging is enabled in Asterisk. Manager
      sessions will receive manager events for each granular fax session
      event. This is an optional argument (default: off)
      'f' - Allow fax fallback to audio mode on T.38-capable channels
      's' - Send progress Manager events (overrides statusevents setting in
      res_fax.conf)
```

The following input channel variables are used by ReceiveFAX for backwards compatibility with previous spandsp-based Asterisk applications. New development and dialplan creators should use the FAXOPT dialplan function. FAXOPT will override channel variables in the slim case that both FAXOPT and a channel variable were used for the same call to SendFAX or ReceiveFAX; *e.g.* FAXOPT(headerinfo) and LOCALHEADERINFO.

- **LOCALSTATIONID** – Text string that identifies the sender identification to the remote side of the fax transmission.
- **LOCALHEADERINFO** – Text string that becomes the fax header sent on each page. If this variable is not set, no header will be used.

The 1.4 version of res_fax supports the following channel variables for backwards compatibility, but it should be noted that these channel variables are deprecated and not supported in 1.6 or newer versions of res_fax.

- **FAX_DISABLE_V17** – Set to '1' to disable V.17.
- **FAX_FORCE_V17** – Set to '1' to force V.17.
- **FAX_FORCE_V27** – Set to '1' to force V.27.
- **FAX_FORCE_V29** – Set to '1' to force V.29.
- **FAX_FORCE_V34** – Set to '1' to force V.34.
- **PHASEESTATUS** – This channel variable will always have the same value as the FAXSTATUS channel variable and is in only intended to ease conversion to res_fax.
- **PHASEESTRING** – This channel variable will always have the same value as the FAXSTATUSSTRING channel variable and is only intended to ease conversion to res_fax.

The following output variables are set by ReceiveFAX when the fax transmission completes.

- **FAXSTATUS** – The fax operation result.
- **FAXERROR** – The reason for a fax failure.
- **FAXSTATUSSTRING** – The fax operation result string.
- **REMOTESTATIONID** – Text string that identifies the remote station.
- **FAXPAGES** – The number of pages transferred during the fax transmission.
- **FAXBITRATE** – The transmission rate used for the fax transmission.
- **FAXRESOLUTION** – The fax image resolution used for the fax transmission.

3.2 res_fax.conf

The res_fax.conf file is optional and will support the configuration options listed in the table below. If the res_fax.conf is not found at module load time, compile-time defaults will be used. The res_fax module reads the *[general]* section of res_fax.conf. In addition to the configuration file, refer to section *3.1.1* for ways to modify configuration settings via the dialplan or per-call operations.

Parameter	Section	Definition	Values	Default
ecm	general	Error Correction Mode (ECM) for G.711 fax sessions	yes | no	yes
minrate	general	Minimum fax transmission rate	2400 | 4800 | 7200 | 9600 | 12200 | 14400	2400
maxrate	general	Maximum fax transmission rate	2400 | 4800 | 7200 | 9600 | 12200 | 14400	14400
rxtxappnames	general	Use “RxFAX”/”TxFAX” application names instead of “ReceiveFAX”/”SendFAX” for the 1.4 version of res_fax.	yes | no	no
statusevents	general	Enable reporting of fax transmission status events to manager sessions with 'call' class permissions	yes | no	no

3.3 res_fax_digium.conf

The res_fax_digium.conf file is optional and will support the configuration option(s) listed in the table below. If the res_fax_digium.conf is not found at module load time, compile-time defaults will be used. The res_fax_digium module reads the *[general]* section of res_fax_digium.conf. In addition to the configuration file, refer to section *3.1.1* for ways to modify configuration settings via the dialplan or per-call operations.

Parameter	Section	Definition	Values	Default
maxdelay	general	Maximum expected T.38 delay is a measure in milliseconds and is used to determine the default size of T.38 packets sent to/from the fax stack	<int>	800

> **Note:** The maxdelay T.38-affecting parameter listed in the table above is not available in the Open Source Asterisk 1.4 releases. It is available in the Open Source Asterisk 1.6 (or later) releases.

3.4 Compatibility with spandsp

Efforts were taken to make res_fax backwards compatible with previous Asterisk fax applications based on spandsp. Sections *4.4* and *4.4* provide information that may help a spandsp-based fax user move to res_fax, or vice versa.

Chapter 4: Troubleshooting

This chapter provides various methods for obtaining the necessary information to troubleshoot most problems relating to Digium's Fax For Asterisk. Multiple resources are available to obtain more information about Asterisk and Digium products. These resources are listed on page 38.

4.1 Manager Events

The fax modules are capable of sending various types of manager events to manager sessions that are capable of receiving *CALL* class manager events. The manager events listed below are sent by the fax modules and detailed in this section.

- Fax Transmission Completion Events
- Fax Status Events
- T.38 Fax Status Events
- Fax Document Status Events

4.1.1 Fax Transmission Completion Events

One fax transmission completion event is always sent at the end of each fax transmission, regardless of the outcome of the fax attempt. An example SendFAX transmission completion manager event is listed below.

```
SendFAX
Channel:            <channel>
Exten:              <extension>
CallerID:           <caller identification>
RemoteStationID:    <remote station identification>
LocalStationID:     <local station identification>
PagesTransferred:   <pages transferred>
Resolution:         <negotiated resolution>
TransferRate:       <negotiated rate>
```

```
FileName:            <filename>
```

Note: The ReceiveFAX event is identical, except for the event name.

4.1.2 Fax Status Events

Fax status events are sent to manager sessions when the *statusevents* configuration file option is enabled. Below is an example SendFAX status message.

```
SendFAXStatus
Status:         <status>
Channel:        <channel>
Exten:          <extension>
CallerID:       <caller identification>
LocalStationID: <local station identification>
FileName:       <file>
```

Note: The ReceiveFAXStatus event is identical, except for the event name.

A description of the status messages and message purpose is listed below.

- **Allocating Resources** – Status message sent prior to the creation of the fax session.
- **No Available Resource** – Status message sent if the system failed to create a fax session. Possible reasons for this message would be:
  1. Request to create fax session exceeds the maximum number of licensed fax channels.
  2. No fax technology module registered with res_fax has the required capabilities to fulfill the fax session request.
  3. System resource limitations prevented the allocation of system resources.
- **Negotiating T.38** – Status message indicating a request to negotiate T.38 has been sent from the fax application to the channel driver. If a channel driver cannot support T.38, this request will be ignored and no event will be returned to the application. **This message is not used with the Open Source Asterisk 1.4 releases.**

- **T.38 Negotiated** – Status message indicating a successful negotiation of T.38 and the creation of a T.38-capable fax session. **This message is not used with the Open Source Asterisk 1.4 releases.**
- **Starting Fax Transmission** – Status message indicating that successful initiation of the fax session. No more 'status' messages will be sent for the remainder of the fax transmission, but a completion event is always sent to manager sessions regardless of the *statusevents* configuration option.

4.1.3 T.38 Fax Status Events

T.38 fax status events are sent to manager sessions when the 'd' debug application argument is specified. T.38 Fax Status Events are not supported with the Open Source Asterisk 1.4 releases. Below is an example T.38 fax status event.

```
T38FaxStatus
Channel:                  <channel>
Fax Session:              <fax session identification>
Max Lag:                  <max lag in ms>
Total Lag:                <total lag in ms>
Average Lag:              <average lag>
Total Events:             <total T.38 events>
T38 Session Duration:     <session duration in sec>
T38 Packets Sent:         <num packets sent>
T38 Octets Sent:          <num octets sent>
Average Tx Data Rate:     <average rate>
T38 Packets Received:     <num packets received>
T38 Octets Received:      <num octets received>
Average Rx Data Rate:     <average received>
Jitter Buffer Overflows:  <overflows>
Minimum Jitter Space:     <min buffer space>
Unrecovered Packets:      <unrecovered packets>
```

4.1.4 Fax Document Status Events

Fax document status events are sent to manager sessions when the 'd' debug application argument is specified. Below is an example of a fax document status event.

```
FaxDocumentStatus
Channel:                <channel>
Fax Session:            <fax session identification>
Document Number:        <doc number>
Processed Status:       <status>
Last Error:             <last error>
Page Count:             <page count>
Start Page:             <start page>
Last Page Processed:    <last page>
Retransmission Count:   <retransmission count>
Local NSF Length:       <local NSF length>
Remote NSF Length:      <remote NSF length>
Transfer PELS:          <transfer pels>
Transfer Rate:          <rate>
Transfer Duration:      <duration>
Bad Line Count:         <bad lines>
Document Time:          <document time>
Local SID:              <local SID>
Local NSF:              <local NSF>
Local DIS:              <local DIS>
Remote SID:             <remote SID>
Remote NSF:             <remote NSF>
Remote DIS:             <remote DIS>
```

4.2 Manager Actions

The manager actions listed below are provided by the fax modules and detailed in this section.

- FaxLicenseList
- FaxLicenseSttus

4.2.1 FaxLicenseList Action

Issuing the FaxLicenseList AMI action will display all Fax For Asterisk licenses and their loading status. Below is an example manager action using FaxLicenseList.

```
Action: FaxLicenseList

Response: Success
Message: License list will follow

Event: FaxLicense
File: FAX-EXAMPLE1.lic
Key: FAX-EXAMPLE1
Product: RESFAX
Host-ID: ex:am:pl:e0:ex:am:pl:e0:ex:am:pl:e0:ex:am:pl:e0:ex:am:pl:e0
Ports: 200
Status: OK

Event: FaxLicenseList complete
```

4.2.2 FaxLicenseStatus Action

Issuing the FaxLicenseStatus AMI action will display Fax For Asterisk license utilization. Below is an example manager action using FaxLicenseStatus.

```
Action: FaxLicenseStatus

Response: Success
PortsLicensed: 200
```

4.3 Asterisk Command Line Interface (CLI)

The Asterisk CLI provides the operations in the list below:

- fax set debug on
- fax set debug off
- fax set g711cap off
- fax set g711cap on
- fax set t38cap off
- fax set t38cap on
- fax show capabilities
- fax show hostid
- fax show licenses
- fax show session <id>
- fax show sessions
- fax show settings
- fax show stats
- fax show version

4.3.1 fax set debug on

This CLI operation enables fax debugging on all sessions created after this operation is used. Sessions that are already active when this CLI operation was executed will not have debugging enabled unless the 'd' application argument was used. Fax debugging results in the extra manager events described in section *4.1*.

The res_fax_digium module logs granular fax events when system verbosity is greater than '4'.

G.711 Fax sessions will also have frame payloads scanned for silence/energy in the direction of channel-to-stack and stack-to-channel. This output has been very useful for the detection of audio underruns and/or gaps of silence in the audio stream that cause faxes to fail due to carrier loss. Payload scanning results are only logged when verbosity is greater than '5'.

4.3.2 fax set debug off

This CLI operation disables fax debugging on all sessions created after this operation is executed. The only sessions that will have debugging enabled after this operation is executed are fax sessions that are started with the 'd' application argument.

4.3.3 fax set g711cap off

This CLI operation disables the creation of audio capture files for G.711 fax sessions.

4.3.4 fax set g711cap on

This CLI operation enables the creation of audio capture files for G.711 fax sessions. Each session will be stored in a file named with the channel's unique ID and located in the 'g711cap' subdirectory of the Asterisk log directory (set via astlogdir in asterisk.conf). The file will be a stereo WAV file in signed linear (8 KHz sample rate, 16-bit samples) with the left channel being the audio from the remote endpoint and the right channel being the audio from Asterisk.

4.3.5 fax set t38cap off

This CLI operation disables the creation of packet capture files for T.38 fax sessions.

4.3.6 fax set t38cap on

This CLI operation enables the creation of packet capture files T.38 fax sessions. Each session will be stored in a file named with the channel's unique ID and located in the 't38cap' subdirectory of the Asterisk log directory (set via astlogdir in asterisk.conf). The packets in the capture file will appear to be between two endpoints at the IP address 127.0.0.1, with packets from the remote endpoint sent to Asterisk originating from port '1' and packets from Asterisk originating from port '2'.

4.3.7 fax show capabilities

This CLI operation displays the “Type” and “Description” for all registered fax technology modules when this operation is executed.

4.3.8 fax show hostid

This CLI operation displays the Fax For Asterisk Host-ID.

4.3.9 fax show licenses

This CLI operation displays the Fax For Asterisk licensing information.

4.3.10 fax show session <id>

This CLI operation displays detailed information about a fax session identified by its fax session id.

4.3.11 fax show sessions

This CLI operation displays basic information about all the current fax sessions. This basis information includes the channel, technology type, Fax ID, fax type, operation mode, current state, and filename(s) for each fax session.

4.3.12 fax show settings

This CLI operation displays the global settings and defaults of both the Fax core and technology modules.

4.3.13 fax show stats

This CLI operation displays general statistics about fax attempts, successes, and failures.

4.3.14 fax show version

This CLI operation displays the version of the fax modules which are loaded.

4.4 Frequently Asked Questions

This section provides frequently asked questions and resolutions as identified by Digium Technical Support and Engineering.

Which configuration files must be modified?

The /etc/asterisk/res_fax.conf and /etc/asterisk/res_fax_digium.conf are optional configuration files. They only needs to be used when the compile-time default settings need to be changed.

The /etc/asterisk/chan_dahdi.conf file needs to be modified if *faxdetect* functionality is required.

The /etc/asterisk/sip.conf file should be modified to enable T.38 or *faxdetect* functionality. To enable T.38 support, uncomment *'t38pt_udptl = yes'*. To enable *faxdetect* functionality, uncomment *'faxdetect = yes'*.

> **Note:** SIP *faxdetect* functionality is available only in Asterisk 1.6 and later.

To modify UDPTL settings (used in T.38 negotiation), modify the /etc/asterisk/udptl.conf file.

Should I add a load line for res_fax and/or res_fax_digium to my /etc/asterisk/modules.conf file?

It is not required or recommended to specify a load line in the /etc/asterisk/modules.conf for the res_fax.so or res_fax_digium.so files. Asterisk will automatically load them using the autoload option. The autoload option is enabled by default.

Where can I find knowledge base articles for Fax For Asterisk?

Please visit the Fax For Asterisk category of the Digium Knowledge Base:
http://kb.digium.com/?CategoryID=263

Why can't I send or receive T.38 faxes using the res_fax modules with an Open Source Asterisk 1.4 release?

The Open Source Asterisk 1.4 releases do not support T.38 Termination. If you need T.38 Termination support with an Open Source Asterisk release, you must use an Open Source

Asterisk 1.6 (or later) release.

If I am using an Open Source Asterisk 1.4 release, which version of Open Source Asterisk is recommended?

Digium recommends using Open Source Asterisk 1.4.22 or newer. Versions prior to 1.4.22 have not been tested.

If I am faxing over TDM, which version of DAHDI is recommended?

Digium recommends using DAHDI 2.1.0.3 or newer. Versions prior to 2.1.0.3 have not been tested.

What are the compatibility differences between res_fax and spandsp-based modules?

See sections *4.4* and *4.4* for more information.

How do I create a TIFF file?

There are many tools to create a TIFF file. One of the most common is the ghostscript utility, available from *http://www.ghostscript.com*.

Using the command-line ghostscript utility, a Letter-size (8.5" x 11") PDF can be converted to a TIFF file using the following command:

```
# gs -q -dNOPAUSE -dBATCH -sDEVICE=tiffg4 -sPAPERSIZE=letter
-sOutputFile=<dest.tiff> <src.pdf>
```

To create a TIFF from an A4-size (210mm x 297mm) PDF file, use the following command:

```
# gs -q -dNOPAUSE -dBATCH -sDEVICE=tiffg4 -sPAPERSIZE=a4
-sOutputFile=<dest.tiff> <src.pdf>
```

When PDF files are created by document scanners, they are sometimes created with a larger-than-standard paper size, e.g. 8.6" x 12". In these cases, ghostscript does not adjust the size to a Standard (Letter or A4), even if PAPERSIZE is specified. This will cause SendFAX to fail with the following error:

```
ERROR[31106]: res_fax_digium.c:2114 dgm_fax_start: FAX handle 0: failed to
queue document 'document name'
```

To prevent this, the size of the TIFF file needs to be specified in pixels. The following command will create TIFF files with a correct width and length:

For Letter-size paper (8.5" x 11"):

```
# gs -q -dNOPAUSE -dBATCH -sDEVICE=tiffg4 -sPAPERSIZE=letter
-g1728x2150 -sOutputFile=<dest.tiff> <src.pdf>
```

For A4-size paper (210mm x 297mm):

```
# gs -q -dNOPAUSE -dBATCH -sDEVICE=tiffg4 -sPAPERSIZE=a4 -g1680x2285
-sOutputFile=<dest.tiff> <src.pdf>
```

Note: Use of the -g option with PDF files smaller in size than either Letter or A4 should be avoided as its use will enlarge smaller PDFs.

Why can't I send T.38 faxes?

Be sure that you modified /etc/asterisk/sip.conf and uncommented 't38pt_udptl = yes' because this option is disabled by default.

Why are my G.711 faxes getting canceled?

By default, the fax applications set up G.711 fax sessions. If T.38 is negotiated during a G.711 fax session, the G.711 fax session will be canceled and a new T.38 fax session will take over the fax transmission. The only exception is when T.38 has already been successfully negotiated before the fax application is called by Asterisk. In this case, a G.711 fax session will have never been created and the T.38 fax session will operate for the life of the fax transmission.

Why are my faxes negotiating T.38 instead of G.711?

There is no way to force G.711 when a SIP peer has UDPTL enabled and the far end also supports T.38. Asterisk will use T.38 instead of G.711 because T.38 is a more reliable form of communication. If you have UDPTL enabled in the [general] section of sip.conf and want to force G.711 for a specific peer, disable UDPTL from that peer's context.

Why do I get “Cannot create fax session – session limit exceeded” when attempting faxes?

You have exceeded the number of allowed Fax channels according to your available licenses. To purchase additional channel licenses, please visit *http://www.digium.com*.

Why do I get “Only one Free Fax For Asterisk channel is allowed. Ignoring additional licenses.” when I load res_fax_digium?

Only one Free Fax For Asterisk channel is allowed per system. Once a Free Fax For Asterisk license is detected, all subsequent Free Fax For Asterisk licenses will be ignored. All Free Fax For Asterisk licenses are limited to 1 channel. If you need more fax channels, you must purchase additional channels from Digium and register the purchased Fax For Asterisk key to activate the additional channels.

Can the app_fax.so and res_fax.so Asterisk modules be loaded at the same time?

No. If you attempt to load both of them at the same time, Asterisk will report the following:

```
WARNING[XXXXX]: pbx.c:XXXX ast_register_application2: Already have an
application 'SendFAX'
WARNING[XXXXX]: pbx.c:XXXX ast_register_application2: Already have an
application 'ReceiveFAX'
```

The recommended solution for this problem is to edit the */etc/asterisk/mdoules.conf* file to explicitly prevent the app_fax.so Asterisk module from loading. An example is provided below.

```
noload => app_fax.so
```

My res_fax_digium.so fails to load with "Error loading module 'res_fax_digium.so': /usr/lib/asterisk/modules/res_fax_digium.so: cannot restore segment prot after reloc: Permission denied". How do I resolve this?

Disable SELinux using the steps below.

1. Edit the */etc/selinux/config* file.
2. Set SELINUX=disabled.
3. Reboot.

What details should I submit to Technical Support when I am having fax problems?

Support is only provided for customers of Fax For Asterisk.

For G.711 fax issues, perform the following steps:

1. At the Asterisk CLI, type “fax set debug on”.

2. At the Asterisk CLI, type “core set verbose 6”. Verbosity can be 6 or higher.
3. At the Asterisk CLI, type “fax show settings”.
4. At the Asterisk CLI, type “fax show version”.
5. At a command prompt, type “dahdi_monitor <channel> -r <rx audio file> -t <tx audio>”.
6. Redirect a manager session (with *call* class permissions) to a file.
7. Reproduce the issue.
8. Submit Asterisk CLI output, dahdi_monitor recordings, and manager session output to Support.

For T.38 fax issues, perform the following steps:

1. At the Asterisk CLI, type “fax set debug on”.
2. At the Asterisk CLI, type “core set verbose 6”. Verbosity can be 6 or higher.
3. At the Asterisk CLI, type “fax show settings”.
4. At the Asterisk CLI, type “fax show version”.
5. Redirect a manager session (with *call* class permissions) to a file.
6. Reproduce the issue.
7. Submit Asterisk CLI and manager session output to Support.

Where can customers of Fax For Asterisk find answers to additional questions?

There are several places to inquire for more information about Asterisk Digium products:

Digium Technical Support (+1.256.428.6161), or Toll Free in the U.S. (1.877.344.4861), is available 7am-8pm Central Time (GMT -6), Monday - Friday.

Asterisk users mailing list (*www.asterisk.org*, *lists.digium.com*)

IRC channel **#asterisk** on (irc.freenode.net)

Subscription Services Program

Digium is dedicated to supporting your Asterisk system by offering full technical support through our Subscription Services Program. Through this program, you can be at ease knowing that your business will always have access to the Asterisk experts. Pricing on Subscription Services may be obtained from your nearest reseller or you may call Digium Sales for referral to your nearest reseller at +1.256.428.6000 or e-mail sales@digium.com.

Appendix A: Dialplan Examples

A.1 Simple Fax Transmit / Receive

The following is a dialplan example for a simple fax transmit and receive.

```
[fax-rx]
exten => receive,1,NoOp(**** FAX RECEIVE ****)
exten => receive,n,Set(GLOBAL(FAXCOUNT)=$[ ${GLOBAL(FAXCOUNT)} + 1 ])
exten => receive,n,Set(FAXCOUNT=${GLOBAL(FAXCOUNT)})
exten => receive,n,Set(FAXFILE=fax-${FAXCOUNT}-rx.tif)
exten => receive,n,Set(GLOBAL(LASTFAXCALLERNUM)=${CALLERID(num)})
exten => receive,n,Set(GLOBAL(LASTFAXCALLERNAME)=${CALLERID(name)})
exten => receive,n,NoOp(**** SETTING FAXOPT ****)
exten => receive,n,Set(FAXOPT(ecm)=yes)
exten => receive,n,Set(FAXOPT(headerinfo)=MY FAXBACK RX)
exten => receive,n,Set(FAXOPT(localstationid)=1234567890)
exten => receive,n,Set(FAXOPT(maxrate)=14400)
exten => receive,n,Set(FAXOPT(minrate)=2400)
exten => receive,n,NoOp(FAXOPT(ecm) : ${FAXOPT(ecm)})
exten => receive,n,NoOp(FAXOPT(headerinfo) : ${FAXOPT(headerinfo)})
exten => receive,n,NoOp(FAXOPT(localstationid) : ${FAXOPT(localstationid)})
exten => receive,n,NoOp(FAXOPT(maxrate) : ${FAXOPT(maxrate)})
exten => receive,n,NoOp(FAXOPT(minrate) : ${FAXOPT(minrate)})
exten => receive,n,NoOp(**** RECEIVING FAX : ${FAXFILE} ****)
exten => receive,n,ReceiveFAX(/home/dwayne/faxin/${FAXFILE})

; Hangup! Print FAXOPTs
exten => h,1,NoOp(FAXOPT(ecm) : ${FAXOPT(ecm)})
exten => h,n,NoOp(FAXOPT(filename) : ${FAXOPT(filename)})
exten => h,n,NoOp(FAXOPT(headerinfo) : ${FAXOPT(headerinfo)})
exten => h,n,NoOp(FAXOPT(localstationid) : ${FAXOPT(localstationid)})
exten => h,n,NoOp(FAXOPT(maxrate) : ${FAXOPT(maxrate)})
exten => h,n,NoOp(FAXOPT(minrate) : ${FAXOPT(minrate)})
exten => h,n,NoOp(FAXOPT(pages) : ${FAXOPT(pages)})
exten => h,n,NoOp(FAXOPT(rate) : ${FAXOPT(rate)})
exten => h,n,NoOp(FAXOPT(remotestationid) : ${FAXOPT(remotestationid)})
exten => h,n,NoOp(FAXOPT(resolution) : ${FAXOPT(resolution)})
```

```
exten => h,n,NoOp(FAXOPT(status) : ${FAXOPT(status)})
exten => h,n,NoOp(FAXOPT(statusstr) : ${FAXOPT(statusstr)})
exten => h,n,NoOp(FAXOPT(error) : ${FAXOPT(error)})

[fax-tx]
exten => send,1,NoOp(**** SENDING FAX ****)
exten => send,n,Wait(6)
exten => send,n,Set(GLOBAL(FAXCOUNT)=$[ ${GLOBAL(FAXCOUNT)} + 1 ])
exten => send,n,Set(FAXCOUNT=${GLOBAL(FAXCOUNT)})
exten => send,n,Set(FAXFILE=dw-faxout.tif)
; Set FAXOPTs
exten => send,n,NoOp(**** SETTING FAXOPT ****)
exten => send,n,Set(FAXOPT(ecm)=yes)
exten => send,n,Set(FAXOPT(headerinfo)=Fax from $
{GLOBAL(LASTFAXCALLERNAME)} at ${GLOBAL(LASTFAXCALLERNUM)} was received.)
exten => send,n,Set(FAXOPT(localstationid)=1234567890)
exten => send,n,Set(FAXOPT(maxrate)=14400)
exten => send,n,Set(FAXOPT(minrate)=2400)
; Send the fax
exten => send,n,NoOp(**** SENDING FAX : ${FAXFILE} ****)
exten => send,n,SendFAX(/home/dwayne/faxout/${FAXFILE},d)

; Hangup! Print FAXOPTs
exten => h,1,NoOp(FAXOPT(ecm) : ${FAXOPT(ecm)})
exten => h,n,NoOp(FAXOPT(filename) : ${FAXOPT(filename)})
exten => h,n,NoOp(FAXOPT(headerinfo) : ${FAXOPT(headerinfo)})
exten => h,n,NoOp(FAXOPT(localstationid) : ${FAXOPT(localstationid)})
exten => h,n,NoOp(FAXOPT(maxrate) : ${FAXOPT(maxrate)})
exten => h,n,NoOp(FAXOPT(minrate) : ${FAXOPT(minrate)})
exten => h,n,NoOp(FAXOPT(pages) : ${FAXOPT(pages)})
exten => h,n,NoOp(FAXOPT(rate) : ${FAXOPT(rate)})
exten => h,n,NoOp(FAXOPT(remotestationid) : ${FAXOPT(remotestationid)})
exten => h,n,NoOp(FAXOPT(resolution) : ${FAXOPT(resolution)})
exten => h,n,NoOp(FAXOPT(status) : ${FAXOPT(status)})
exten => h,n,NoOp(FAXOPT(statusstr) : ${FAXOPT(statusstr)})
exten => h,n,NoOp(FAXOPT(error) : ${FAXOPT(error)})

[default]
exten => fax,1,NoOp(**** FAX DETECTED ****)
exten => fax,n,Goto(fax-rx,receive,1)
```

A.2 Trunk, app_fax, and spandsp 0.0.6

Dialplan compatibility between Open Source Asterisk trunk using res_fax/res_fax_digium and app_fax/spandsp-0.0.6 was tested using the dialplan below.

- **Inconsistencies between applications:**
  - The FAXSTATUSSTRING channel variable is not used by the Open Source Asterisk trunk app_fax module.
  - The res_fax/res_fax_digium modules do not currently support the ReceiveFAX 'c' (caller mode) option and the SendFAX 'a' (calling mode) arguments.

```
exten => 100,1,Wait(1)
exten => 100,n,Answer()
exten => 100,n,Set(GLOBAL(FAXCOUNT)=$[ ${GLOBAL(FAXCOUNT)} + 1 ])
exten => 100,n,Set(FAXCOUNT=${GLOBAL(FAXCOUNT)})
exten => 100,n,Set(FAXFILE=fax-${FAXCOUNT}-rx.tif)
exten => 100,n,Set(LOCALHEADERINFO=Receiving fax number ${FAXCOUNT})
exten => 100,n,Set(LOCALSTATIONID=${FAXCOUNT})
exten => 100,n,ReceiveFAX(${GLOBAL(FAXRXDIR)}/${FAXFILE})

exten => 111,1,Wait(1)
exten => 111,n,Answer()
exten => 111,n,Set(GLOBAL(FAXCOUNT)=$[ ${GLOBAL(FAXCOUNT)} + 1 ])
exten => 111,n,Set(FAXCOUNT=${GLOBAL(FAXCOUNT)})
exten => 111,n,Set(LOCALHEADERINFO=Sending fax number ${FAXCOUNT})
exten => 111,n,Set(LOCALSTATIONID=${FAXCOUNT})
exten => 111,n,Set(NUMPAGES=${RAND(1,3)})
exten => 111,n,GotoIf($[${NUMPAGES} = 1]?send1page:)
exten => 111,n,GotoIf($[${NUMPAGES} = 2]?send11pages:)
exten => 111,n,GotoIf($[${NUMPAGES} = 3]?send20pages:)
exten => 111,n(send20pages),Set(FAXFILE=${GLOBAL(FAX20PAGES)})
exten => 111,n,Goto(sendit)
exten => 111,n(send11pages),Set(FAXFILE=${GLOBAL(FAX11PAGES)})
exten => 111,n,Goto(sendit)
exten => 111,n(send1page),Set(FAXFILE=${GLOBAL(FAX1PAGE)})
exten => 111,n(sendit),SendFAX(${GLOBAL(FAXTXDIR)}/${FAXFILE})

exten => h,1,NoOp(FaxStatus : ${FAXSTATUS})
exten => h,n,NoOp(FaxStatusString : ${FAXSTATUSSTRING})
exten => h,n,NoOp(FaxError : ${FAXERROR})
exten => h,n,NoOp(RemoteStationID : ${REMOTESTATIONID})
```

```
exten => h,n,NoOp(FaxPages : ${FAXPAGES})
exten => h,n,NoOp(FaxBitRate : ${FAXBITRATE})
exten => h,n,NoOp(FaxResolution : ${FAXRESOLUTION})
```

A.3 Asterisk 1.4, agx-ast-addons, and spandsp 0.0.4

Dialplan compatibility of Asterisk 1.4 dialplans using agx-ast-addons with spandsp was tested.

- **Inconsistencies between applications:**
  - The app_rxfax and app_txfax modules register “RxFAX” and “TxFAX” instead of “ReceiveFAX” and “SendFAX”. To ease the conversion from spandsp-based fax applications to res_fax on 1.4 versions of Asterisk *rxtxappnames=yes* can be specified in /etc/asterisk/res_fax.conf. This configuration option will register “RxFAX” and “TxFAX” application names instead of “ReceiveFAX” and “SendFAX”.
  - The app_rxfax and app_txfax modules use PHASEESTATUS and PHASEESTRING channel variables. These channel variables were added to the 1.4 version of res_fax, and deprecated, to ease the conversion to res_fax. The 1.4 versions of res_fax will set PHASEESTATUS and PHASEESTRING to the same value as the FAXSTATUS and FAXSTATUSSTRING channel variables. The 1.4 versions of res_fax will create all 4 channel variables (PHASEESTATUS, PHASEESTRING, FAXSTATUS, and FAXSTATUSSTRING) before the fax applications exit. The recommended dialplan modification is to move away from channel variables in favor of the FAXOPT dialplan function. The example below illustrates replacement of these channel variables.

    ```
    exten => h,n,NoOp(FAXOPT(status) : ${FAXOPT(status)})
    exten => h,n,NoOp(FAXOPT(statusstr) : ${FAXOPT(statusstr)})
    ```

  - The res_fax/res_fax_digium modules do not currently support the RxFAX 'c' (caller mode) option and the TxFAX 'a' (calling mode) arguments.

The following is a list of components used for this comparison.

- Asterisk 1.4.22
- agx-ast-addons
    - svn URL: https://agx-ast-addons.svn.sourceforge.net/svnroot/agx-ast-addons
    - revision 40
- spandsp-0.0.4pre16

Appendix B: Glossary and Acronyms

ANSI *American National Standards Institute*

An organization which proposes and establishes standards for international communications.

asynchronous

Not synchronized; not timed to an outside clock source. Transmission is controlled by start bits at the beginning and stop bits at the end of each character. Asynchronous communications are often found in internet access and remote office applications.

attenuation

The dissipation of a transmitted signal's power as it travels over a wire.

bandwidth

The capacity to carry traffic. Higher bandwidth indicates the ability to transfer more data in a given time period.

bit

The smallest element of information in a digital system. A bit can be either a zero or a one.

bps *bits per second*

A measurement of transmission speed across a data connection.

broadband

Broadband transmission shares the bandwidth of a particular medium (copper or fiber optic) to integrate multiple signals. The channels take up different frequencies on the cable, integrating voice, data, and video over one line.

channel

A generic term for an individual data stream. Service providers can use multiplexing techniques to transmit multiple channels over a common medium.

Cat5

Category of Performance for wiring and cabling. Cat 5 cabling support applications up to 100 MHz.

Cat5E

Category of Performance for wiring and cabling. Category 5 Enhanced wiring supports signal rates up to 100 MHz but adheres to stricter quality specifications.

CLEC *competitive local exchange carrier*

A term for telephone companies established after the Telecommunications Act of 1996 deregulated the LECs. CLECs compete with ILECs to offer local service. See also LEC and ILEC.

CO *central office*

The CO houses local switching equipment. All local access lines in a particular geographic area terminate at this facility (which is usually owned and operated by an ILEC).

CPE *customer premises equipment*

Terminal equipment which is connected to the telecommunications network and which resides within the home or office of the customer. This includes telephones, modems, terminals, routers, and television set-top boxes.

DAHDI *Digium Asterisk Hardware Device Interface*

A telephony project dedicated to implementing a reasonable and affordable computer telephony platform into the world marketplace. Also, the collective name for the Digium-provided drivers for Digium telephony interface products.

DS0 *Digital Signal, Level 0*

A voice grade channel of 64 Kbps. The worldwide standard speed for digitizing voice conversation using PCM (Pulse Code Modulation).

DS1 *Digital Signal, Level 1*

1.544 Mbps in North America (T1) and Japan (J1) -up to 24 voice channels (DS0s), 2.048 Mbps in Europe (E1) - up to 32 voice channels (DS0s). DS1/T1/E1 lines are part of the PSTN.

DS3 *Digital Signal, Level 3*

T3 in North America and Japan, E3 in Europe. Up to 672 voice channels (DS0s). DS3/T3/E3 lines are not part of the PSTN

DTMF *Dual Tone Multi-Frequency*

Push-button or touch tone dialing.

E1

The European equivalent of North American T1, transmits data at 2.048 Mbps, up to 32 voice channels (DS0s).

E3

The European equivalent of North American T3, transmits data at 34.368 Mbps, up to 512 voice channels (DS0s). Equivalent to 16 E1 lines.

ECM *Error Correction Mode*

EMI *Electromagnetic Interference*

Unwanted electrical noise present on a power line.

Ethernet

Ethernet is a family of frame-based computer networking technologies for local area networks (LANs). It defines a number of wiring and signaling standards for the Physical Layer of the OSI networking model, through means of network access at the Media Access Control (MAC) / Data Link Layer, and a common addressing format.

full duplex

Data transmission in two directions simultaneously.

FXO *Foreign Exchange Office*

Receives the ringing voltage from an FXS device.

FXS *Foreign Exchange Station*

Initiates and sends ringing voltage.

G.711

A recommendation by the Telecommunication Standardization Sector (ITU-T) for an algorithm designed to transmit and receive mulaw PCM voice and A-law at a digital bit rate of 64 Kbps. This algorithm is used for digital telephone sets on digital PBX.

G.723.1

A recommendation by the Telecommunication Standardization Sector (ITU-T) for an algorithm designed to transmit and receive audio over telephone lines at 6.3 Kbps or 5.3 Kbps.

G.729a

A recommendation by the Telecommunication Standardization Sector (ITU-T) for an algorithm designed to transmit and receive audio over telephone lines at 8 Kbps.

H.323

A recommendation by the Telecommunication Standardization Sector (ITU-T) for multimedia communications over packet-based networks.

half duplex

Data transmission in only one direction at a time.

IAX *Inter-Asterisk eXchange*

The native VoIP protocol used by Asterisk. It is an IETF standard used to enable VoIP connections between Asterisk servers, and between servers and clients that also use the IAX protocol.

ILBC *internet Low Bitrate Codec*

A free speech codec used for voice over IP. It is designed for narrow band speech with a payload bitrate of 13.33 kbps (frame length = 30ms) and 15.2 kbps (frame length = 20 ms).

ILEC *incumbent local exchange carrier*

The LECs that were the original carriers in the market prior to the entry of competition and therefore have the dominant position in the market.

interface

A point of contact between two systems, networks, or devices.

ISO *International Standards Organization*

LED *light-emitting diode*

Linux

A robust, feature-packed open source operating system based on Unix that remains freely available on the internet. It boasts dependability and offers a wide range of compatibility with hardware and software. Asterisk is supported exclusively on Linux.

loopback

A state in which the transmit signal is reversed back as the receive signal, typically by a far end network element.

MAC address *Media Access Control address*

A quasi-unique identifier assigned to most network adapters or network interface cards (NICs) by the manufacturer for identification.

MGCP *Media Gateway Control Protocol*

multiplexing

Transmitting multiple signals over a single line or channel. FDM (frequency division multiplexing) and TDM (time division multiplexing) are the two most common methods. FDM separates signals by dividing the data onto different carrier frequencies, and TDM separates signals by interleaving bits one after the other.

MUX *multiplexer*

A device which transmits multiple signals over a single communications line or channel. See multiplexing.

open source

Software distributed as source code under licenses guaranteeing anybody rights to freely use, modify, and redistribute the code.

OSI Reference Model *Open Systems Interconnection Reference Model*

An abstract description for layered communications and computer network protocol design.

packet

A formatted unit of data carried by a packet mode computer network.

PBX *private branch exchange*

A smaller version of a phone company's large central switching office. Example: Asterisk.

PCI *peripheral component interconnect*

A standard bus used in most computers to connect peripheral devices.

PDF *Portable Document Format*

A file format created by Adobe Systems Incorporated for document exchange. PDF is used for representing two-dimensional documents in a manner independent of the application software, hardware, and operating system.

POP *point of presence*

The physical connection point between a network and a telephone network. A POP is usually a network node serving as the equivalent of a CO to a network service provider or an interexchange carrier.

POTS *plain old telephone service*

Standard phone service over the public switched telephone network (PSTN). This service provides analog bandwidth of less than 4 kHz.

PPP *point-to-point protocol*

Type of communications link that connects a single device to another single device, such as a remote terminal to a host computer.

PSTN *public switched telephone network*

The public switched telephone network (PSTN) is the network of the world's public circuit-switched telephone networks. Originally a network of fixed-line analog telephone systems, the PSTN is now almost entirely digital, and now includes mobile as well as fixed telephones.

QoS *quality of service*

A measure of telephone service, as specified by the Public Service Commission.

RJ11

A six-pin jack typically used for connecting telephones, modems, and fax machines in residential and business settings to PBX or the local telephone CO.

SIP *Session Initiation Protocol*

An IETF standard for setting up sessions between one or more clients. It is currently the leading signaling protocol for Voice over IP, gradually replacing H.323.

source code

Any collection of statements or declarations written in some human-readable computer programming language.

T.30

A recommendation by the Telecommunication Standardization Sector (ITU-T) for Group 3 fax machines that specifies the handshaking, protocols, and error correction. T.4 and T.30 make up the complete standard for Group 3 fax.

T.38

A recommendation by the Telecommunication Standardization Sector (ITU-T) to permit faxes to be transported across IP networks between existing Group 3 fax terminals in real time.

T.4

A recommendation by the Telecommunication Standardization Sector (ITU-T) for Group 3 fax machines that specifies the page dimensions, resolutions, and compression scheme. T.4 and T.30 make up the complete standard for Group 3 fax.

T1

A dedicated digital carrier facility which transmits up to 24 voice channels (DS0s) and transmits data at 1.544 Mbps. Commonly used to carry traffic to and from private business networks and ISPs.

T3

A dedicated digital carrier facility which consists of 28 T1 lines and transmits data at 44.736 Mbps. Equivalent to 672 voice channels (DS0s).

TDM *time division multiplexer*

A device that supports simultaneous transmission of multiple data streams into a single high-speed data stream. TDM separates signals by interleaving bits one after the other.

telco

A generic name which refers to the telephone companies throughout the world, including RBOCs, LECs, and PTTs.

TIFF *Tagged Image File Format*

A file format for storing images.

tip and ring

The standard termination on the two conductors of a telephone circuit; named after the physical appearance of the contact areas on the jack plug.

twisted pair

Two copper wires commonly used for telephony and data communications. The wires are wrapped loosely around each other to minimize radio frequency interference or interference from other pairs in the same bundle.

V *volts*

V.17

A recommendation by the Telecommunication Standardization Sector (ITU-T) that uses TCM modulation at 12,000 and 14,400 bps for Group 3 fax transmissions. It adds TCM to the V.29 standard at 7,200 and 9,600 bps to allow transmission over noisier lines.

V.21

A recommendation by the Telecommunication Standardization Sector (ITU-T) for asynchronous full-duplex communication between two analog dial-up modems using audio frequency-shift keying modulation (FSK) at 300 baud to carry digital data at 300 bit/s. It is a variant of the original Bell 103 modulation format.

V.27ter

A recommendation by the Telecommunication Standardization Sector (ITU-T) for synchronous 2,400 and 4,800 bps half-duplex modems using DPSK modulation on dial-up lines. It includes an optional 75 bps back channel. V.27ter is used in Group 3 fax transmission without the back channel.

V.29

A recommendation by the Telecommunication Standardization Sector (ITU-T) for full-duplex modems allowing synchronous 4,800, 7,200, and 9,600 bps transfer modes (PSK and QAM modulations). It has been adapted for Group 3 fax transmission over dial-up lines at 9,600 and 7,200 bps.

VoIP *Voice over IP*

Technology used for transmitting voice traffic over a data network using the Internet Protocol.

Appendix C: DIGIUM END-USER PURCHASE AND LICENSE AGREEMENT

July 2009

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FAX for Asterisk

HPEC for Asterisk

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