Skype for Asterisk™

Administrator Manual

601-00017 Rev. B3

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Chapter 1: Overview

Digium's Skype for Asterisk™ (SfA) is an add-on channel driver for Asterisk based systems. Adding Skype for Asterisk to any Asterisk server enables complete access to the Skype community, including low cost PSTN access and free calling to over 440+ million Skype users.

Skype for Asterisk integrates seamlessly with the Skype community. Skype for Asterisk performance is superior to the proxy or gateway products available for connecting to the Skype community. There is no secondary piece of hardware to manage as Skype for Asterisk will run directly from an Asterisk-based PBX.

Key Features

- Make Skype to Skype calls
- Calls to landlines and mobile phones
- Receive calls with SkypeIn
- Make worldwide PSTN calls with SkypeOut
- Make and receive multiple concurrent Skype calls from the same Skype account
- Transfer Skype calls
- DTMF support for incoming and outgoing calls
- Read Skype profile fields from incoming calls
- Read Skype Credit balance
- Set and retrieve online status
- Set privacy settings
- Chat with Skype users
- Handle incoming Skype calls using all Asterisk applications (voicemail, ACD, MeetMe conferencing, etc.)

- Simultaneous access from both Asterisk and the Skype desktop client
- Supports G.711 and G.729 (included) codecs

Key Benefits

- Save money with:
    - Free calling to 440+ million Skype users worldwide directly from your Asterisk server
    - Great rates for worldwide inbound calling DIDs via online numbers (SkypeIn)
    - Great rates for worldwide outbound calling to landline and mobile phones (SkypeOut). Please note that Skype for Business subscription prices do not apply.
- Add Skype to your call routing tables to optimize global calling costs
- Add a click to call button to your web site or e-mail so customers can quickly contact you
- Allows customers to call via a local online number
- Perfect for the remote employee as the office is one click away with free calling
- Communicate securely with free, high quality, encrypted Asterisk-to-Asterisk calls.

Skype for Asterisk provides two components: res_skypeforasterisk and chan_skype. The res_skypeforasterisk Asterisk resource module contains the Skype engine, along with various libraries and other components required to talk to the Skype engine and manage user accounts, calls, presence, etc. This module is provided in a binary-only form. The chan_skype Asterisk channel module is the Asterisk channel driver that provides calling services to and from the Skype community, using the library services provided by res_skypeforasterisk.so.

Digium's customers of Skype for Asterisk may purchase license keys coded for a specific number of channels. Each licensed channel allows Skype for Asterisk to initiate a single concurrent call to the Skype community, or to receive a single concurrent call from the Skype community. As customers need to expand their calling 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. If additional channels of Skype capability are required, additional channels of Skype for Asterisk may be purchased from *http://www.digium.com*.

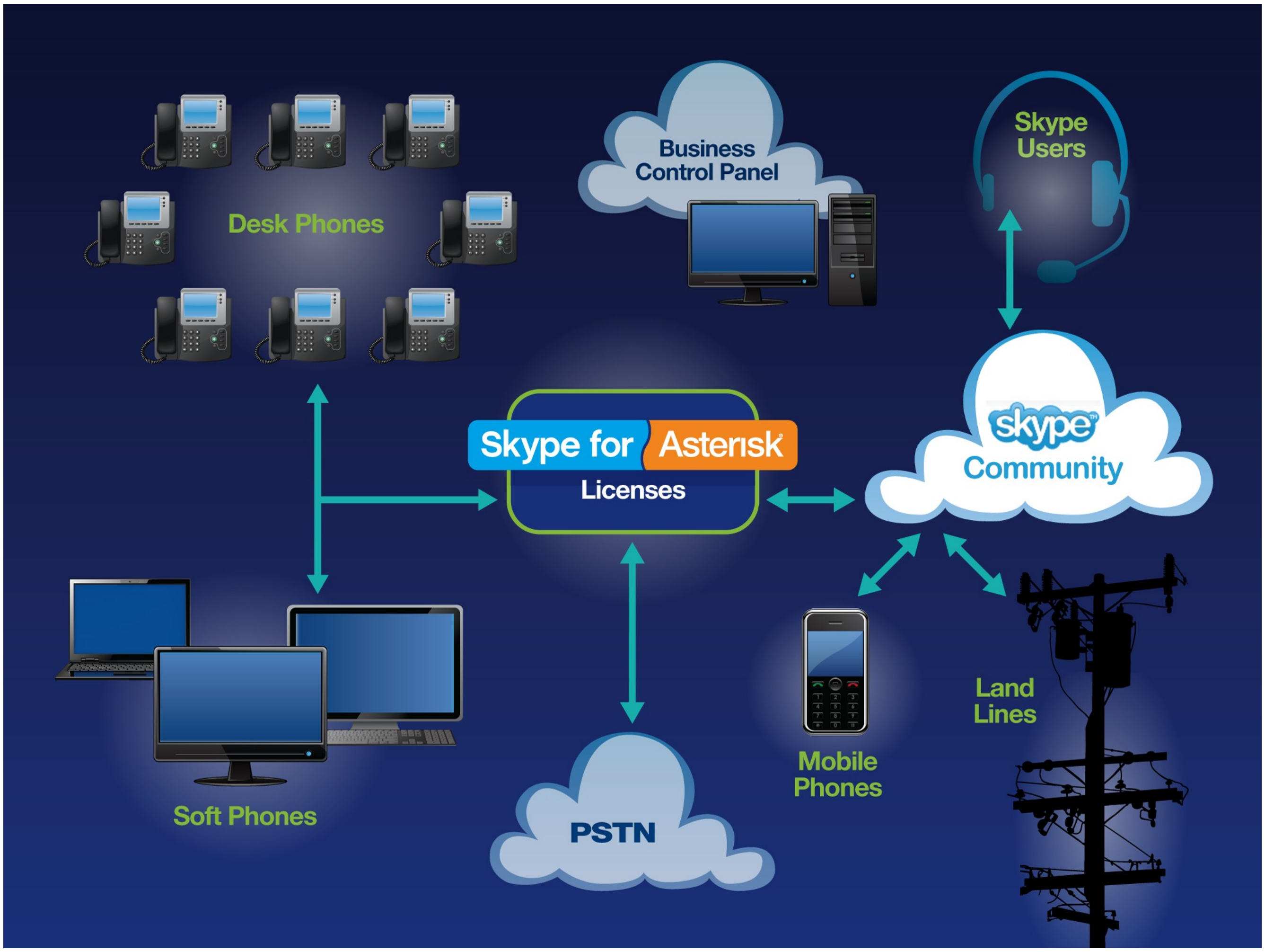

Figure: Skype for Asterisk Application Scenario

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 Skype for Asterisk.

Important Notes:

- Skype for Asterisk is available for Linux only.
- Before installing Skype for Asterisk, it is required that a full installation of Asterisk exist on the system. This includes not only the Asterisk runtime files, but also the development headers. If you installed Asterisk from source and ran “make install”, then you should have all of the required components. If you installed Asterisk from your distribution's package management system (RPM, deb, etc.), then you need to make sure the development library packages are installed as well (asterisk-devel on Red Hat-based systems or asterisk-dev on Debian-based systems).
- 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	1.4.25
Open Source Asterisk branch 1.6.2	1.6.2.0
Open Source Asterisk branch 1.8	1.8.0
AsteriskNOW	1.7

2.1 Installation Overview

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

1. Generate a valid *Skype for Asterisk* license key using the *register* utility.
2. Download and install the *Skype for Asterisk* package that is built for your platform.
3. Install the Digium G.729 software codec that is built for your platform.

   Note: Each *Skype for Asterisk* channel includes a channel license of G.729.

4. Load the res_skypeforasterisk and chan_skype Asterisk modules.

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

The Skype for Asterisk package may be downloaded from:
http://downloads.digium.com/pub/telephony/skypeforasterisk/

> **Note:** Supported software builds are provided for 32-bit and 64-bit x86 platforms. Choose the directory that closest matches your Asterisk version. Each of these directories contains TAR files that include the Skype modules.

2.2 Register Skype for Asterisk

Registration of the Skype 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 Skype for Asterisk license key.

Important Notes:

- Internet access is required from your Asterisk server in order to register your Skype 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 successfully complete.

- Multiple Skype for Asterisk keys may be registered on the same Asterisk server. This will allow you to increase the total number of available Skype for Asterisk channels on your Asterisk server. New Skype for Asterisk keys may be registered to your Asterisk server using the same instructions provided above. There will be an additional Skype for Asterisk license file generated in the /var/lib/asterisk/licenses directory for each Skype for Asterisk key that is successfully registered to your Asterisk server. It is extremely important that you follow the instructions provided in section *2.7* whenever a new Skype for Asterisk key is successfully registered to your Asterisk server.

- A Skype for Asterisk key must be re-registered if any of the Ethernet devices in your Asterisk server are changed, added, or removed. The unique Skype for Asterisk license file that 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 Skype 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 Skype for Asterisk key incremented. Digium reserves the right to deny authorization for having a Skype for Asterisk key incremented.

2.2.1 Open Source Asterisk

An example for 32-bit Linux using Open Source Asterisk is provided below. Be sure to run these commands as the root user.

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

Follow the prompts provided by the registration utility and provide the information it requests to activate your Skype for Asterisk license key.

2.2.2 AsteriskNOW

AsteriskNOW 1.7 systems have the ability to easily download and install the register utility. An example is provided below. Be sure to run these commands as the root user.

```
# yum install register
# register
```

Follow the prompts provided by the registration utility and provide the information it requests to activate your Skype for Asterisk license key.

2.3 Install Skype for Asterisk

There are different versions of Skype for Asterisk that contain both source code and binary-only components for various Asterisk releases; there is a single version for Asterisk 1.4.25 and above, and there are versions for Asterisk 1.6.x point releases (1.6.0, 1.6.1, etc.). The RPM packaged versions of Skype for Asterisk for AsteriskNOW are binary only. Take note that these modules are **not** loadable in prior releases of Asterisk, but will only work with the specific version for which they are designed to be used. Please be sure that you download the correct version of Skype for Asterisk for your Asterisk version.

There are frequently updated builds of Skype for Asterisk posted, and each build has a *version number*. This version number is part of the filename. In this document, build number *1.1.4* is used as an example, but when you read this document the current build number may be different (higher).

2.3.1 Open Source Asterisk

Extract, compile, and install the contents of the Skype for Asterisk package for Open Source Asterisk. An example for 32-bit Linux using Open Source Asterisk branch 1.6.0 is provided below. Be sure to run these commands as the root user.

```
# wget http://downloads.digium.com/pub/telephony/skypeforasterisk/\
  asterisk-1.6.0/x86-32/skypeforasterisk-1.6.0_1.1.4-x86_32.tar.gz
# tar xzvf skypeforasterisk-1.6.0_1.1.4-x86_32.tar.gz
# cd skypeforasterisk-1.6.0_1.1.4-x86_32
# make
# make install
```

If the *chan_skype.conf* file had not been installed from a previous installation of Skype for Asterisk, then the *chan_skype.conf* file will need to be installed by executing the following command. Otherwise, skip this step.

```
# make samples
```

Note: Skype for Asterisk will not properly function if the *chan_skype.conf* file is not installed.

2.3.2 AsteriskNOW

Install the Skype for Asterisk RPM package for AsteriskNOW. An example for 32-bit Linux using AsteriskNOW 1.7 is provided below. Be sure to run these commands as the root user.

```
# yum update asterisk14
# yum install asterisk14-skypeforasterisk
```

If you are upgrading Skype for Asterisk from a previous version on AsteriskNOW, instead of executing "`yum install asterisk14-skypeforasterisk`", use the following command.

```
# yum update asterisk14-skypeforasterisk
```

Note: The FreePBX GUI interface that is provided as part of AsteriskNOW 1.7 is not capable of configuring the Skype for Asterisk product. Skype for Asterisk's product configuration must be managed by direct editing of its configuration file.

2.4 Digium G.729 Software Codec Module

The Digium G.729 software codec module (codec_g729a.so) will need to be installed in addition to *Skype for Asterisk.* Your *Skype for Asterisk* license key may be used to activate a G.729 channel at no additional cost. This requires version 3.0.0 or later of the Digium G.729 software codec module. This allows Skype for Asterisk users to use the G.729 codec for their Skype calls. This is commonly required for SkypeIn and SkypeOut calls. For more information regarding the Digium G.729 software codec module, please read the G.729 README that is available in the documentation section at *http://www.digium.com/support*.

2.5 Load Skype for Asterisk Modules

The res_skypeforasterisk resource module and the chan_skype channel module must be loaded in Asterisk in order to use the Skype for Asterisk channels. There are a few important things that you should know before loading these modules.

The res_skypeforasterisk.so module contains a binary Skype engine called *skyhost*. Skype for Asterisk communicates with *skyhost* to make and manage connections to the Skype community. This engine automatically runs as a separate Linux process called *skypeforasterisk* once chan_skype.so is loaded.

During Skype for Asterisk's initialization process, the engine is extracted into a temporary directory, launched, and then removed. By default, it is extracted into the */tmp* directory. Some Linux distributions mount the */tmp* directory with the *noexec* flag which does not allow files to be executed. If your system is configured to mount the /tmp directory with the noexec flag, the engine_directory configuration option in the *chan_skype.conf* file must be modified to use a directory that will allow the Asterisk process write access and that will allow files to be executed.

The *autoload* option in */etc/asterisk/modules.conf* is enabled by default. As long as you have not disabled it, then the Skype for Asterisk modules will be loaded the next time you start Asterisk. If you have disabled the *autoload* option, then you will need to add the following lines to the bottom of the *[modules]* section of the */etc/asterisk/modules.conf* file.

```
load => res_skypeforasterisk.so
load => chan_skype.so
```

Note: These modules must be loaded in the order provided above.

If Asterisk is already running, you may load the Skype for Asterisk modules from the Asterisk CLI. An example is provided below.

```
*CLI> module load res_skypeforasterisk.so
*CLI> module load chan_skype.so
```

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

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

Reloading this module will only be successful if no Skype calls are in progress. If there are active Skype calls, 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”
```

2.6 Verify Installation

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

```
# asterisk -rvvv

*CLI> skype show licenses
Skype for Asterisk Licensing Information
========================================
Total licensed channels: 100

Licenses Found:
File: S4A-ABCDEFGHIJKL.lic -- Key: S4A-ABCDEFGHIJKL -- Expires: 2039-
07-31 -- Host-ID:
ab:cd:12:34:ab:cd:12:34:ab:cd:12:34:ab:cd:12:34:ab:cd:12:34 --
Channels: 100 (OK)
```

2.7 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 Skype for Asterisk licenses in case you need to reinstall your operating system.

> **Note:** A Skype for Asterisk key must be re-registered if any of the Ethernet devices in your Asterisk server are changed, added, or removed. The unique Skype for Asterisk license file that 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 Skype 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 Skype for Asterisk key incremented. Digium reserves the right to deny authorization for having a Skype for Asterisk key incremented.

Chapter 3: Configuration

Digium's Skype for Asterisk has a variety of configuration options. This chapter provides an explanation of the configuration options that are available.

Important Notes:

- Only accounts created from the Business Control Panel will be usable with Skype for Asterisk. The Skype Business Control Panel is a web-based tool that is free to setup and use. It is accessible from Skype's web site at *http://www.skype.com/business*. All Skype For Asterisk users must be created by clicking on *Add Members*, and then clicking the *Create a business account* button. Inviting users by Skype name or e-mail address is not currently supported.

- The administrator account for the Business Control Panel is a regular Skype account. Due to this fact, the administrator account will not be able to use Skype for Asterisk.

- The current version of chan_skype.so does not provide passthrough G.729 support like other Asterisk channel drivers. When a Skype call wants to use G.729, the codec_g729a.so module must be loaded, and G.729 licensed channels must be available. This will be improved in a future release of Skype for Asterisk. For this release, it is suggested to configure your users to only allow G.711 ulaw and/or alaw in the *chan_skype.conf* file unless you have G.729 licensed channels available.

3.1 chan_skype.conf

The *chan_skype.conf* file is mandatory and is placed in the */etc/asterisk* directory during the installation process. This file documents the configuration options available for the Skype for Asterisk channel driver, including how to define users and log them into the Skype community.

The *general* section contains settings that apply to the entire channel driver and all defined users. The general section appears as *[general]* in the *chan_skype.conf* file.

Parameter	Section	Definition	Values	Default
engine_directory	general	Directory that will be used to hold the Skype engine and its working database. This directory **must** allow executable files to be present and executed.	<directory>	/tmp
default_user	general	Username that will be used for outgoing calls and presence requests if no explicit username is specified.	<username>	none
debug	general	Enable/disable debugging (very verbose)	yes | no	no
bind_address	general	IP address to use for Skype engine	<ip_address>	0.0.0.0 (any)
rtp_address	general	IP address to use for RTP media	<ip_address>	127.0.0.1
bind_port	general	TCP port to use for Skype engine. This setting is only a suggestion to the Skype engine; if it cannot us the specified port, it will automatically fall back to using a random port.	<tcp_port>	0 (random)
disable_tcpauto	general	Disable automatic TCP ports in Skype engine. By default, the Skype engine will listen on a random TCP port or the port specified in 'bindport', and will attempt to listen on ports 80 and 443 (HTTP and HTTPS, respectively). This is done because it usually allows for easier connections through firewalls.	yes | no	no
disable_udp	general	Disable use of UDP in Skype engine. The Skype engine will normally use UDP ports for media streams. In cases where UDP connections cannot or should not be used, this can be disabled.	yes | no	no
https_proxy	general	Enable use of an HTTPS proxy in Skype engine. If your network requires that outbound HTTPS connections be made through a standard HTTPS proxy server, you can specify the proxy server here (either as a hostname or IP address followed by an optional ':' and port number).	<hostname[:port]> | <ip_address[:port]>	none
https_proxy_user	general	Username for HTTPS proxy. If your HTTPS proxy requires a username, it can be specified here.	<username>	none
https_proxy_password	general	Password for HTTPS proxy. If your HTTPS proxy requires a password, it can be specified here.	<password>	none
socks5_proxy	general	Enable use of a SOCKS5 proxy for Skype engine. If your network requires that outbound HTTPS connections be made through a SOCKS5 compatible proxy server, the proxy server can be specified here (either as a hostname or IP address followed by an optional ':' and port number).	<hostname[:port]> | <ip_address[:port]>	none
socks5_proxy_user	general	Username for SOCKS5 proxy. If your SOCKS5 proxy requires a username, it can be specified here.	<username>	none
socks5_proxy_password	general	Password for SOCKS5 proxy. If your SOCKS5 proxy requires a password, it can be specified here.	<password>	none

Each user section identifies a Skype user that the channel driver should log in to the Skype community. The *user* sections appear as *[<username>]* in the *chan_skype.conf* file.

Parameter	Section	Definition	Values	Default
secret	<username>	The user's password	<password>	none
context	<username>	The dialplan context that incoming calls should be directed to for this user.	<dialplan_context>	default
exten	<username>	The extension in the target context that incoming calls should be directed to for this user.	<extension>	<username>
disallow	<username>	The codecs that should be disallowed for calls to/from this user.	<codec>, <codec>, …	all
allow	<username>	The codecs that should be allowed for calls to/from this user.	<codec>, <codec>, …	ulaw, alaw, g729
direction	<username>	Allowed call directions	Incoming | outgoing | both	both
auth_policy	<username>	Incoming buddy list authorization requests. When this user receives a request to authorize being added to another Skype user's buddy list, there are various ways it can be handled: accept (authorize requested), accept.<password> (authorized request if supplied password was sent by requester), deny (deny request), block (deny request and block future requests from the requester), and ignore (ignore request; no response to requester). It is possible to provide multiple values for this setting. They will be processed in the order they are listed; the first match will be used to generate the response.	accept | accept.<password> | deny | block | ignore	accept
buddy_autoadd	<username>	Outgoing buddy list addition requests. When chan_skype receives a presence state request for a Skype user from a dialplan hint or some other mechanism, if that target user is not already on the requesting user's buddy list, then the Skype community will not allow the presence state to be seen. Setting this option will automatically attempt to add the target user to the requesting user's buddy list. If the target user authorizes the request, then future presence state changes for the target user will be received by chan_skype and forwarded into the other Asterisk modules that requested them. A value of *hints* will allow only buddies that receive a presence information request to be added to the buddy list. A value of *buddies* will automatically add all buddies that have been authorized to receive that user's presence information to the buddy list. A value of *'buddies,hints'* will cause both of these to occur.	no | hints | buddies | buddies,hints	no
buddy_presence	<username>	Buddy list presence updates. In some cases, Skype users may have an extremely large number of Skype users on their buddy lists. By default, the channel driver will retrieve presence state and updates for all of these users and pass it into Asterisk. For users with large buddy lists, this could generate a significant amount of load in Asterisk processing presence updates. If there are specific users for which you have no need for buddy presence state information, you can use this option to disable the retrieval and update process.	yes | no	yes
mohinterpret	<username>	Specifies a preference for which music on hold class this channel should listen to when put on hold if the music class has not been set on the channel with Set(CHANNEL(musicclass)=whatever) in the dialplan, and the peer channel putting this channel on hold did not suggest a music class. If this option is set toe “passthrough”, then the hold message will always be passed through as signalling instead of generating hold music locally.	<music_class>	default
mohsuggest	<username>	Specifies which music class to suggest to the peer channel when this channel places the peer on hold.	<music_class>	none

Parameter	Section	Definition	Values	Default
autoreply	<username>	Specifies an autoreply message to be sent whenever the user receives a chat message. This is most useful if an SfA user will not be receiving Skype chat messages and would like to alert the sender that their message is not being received.	<string>	none
webpresence	<username>	Enables or disables the sharing of presence information via Skype web buttons	yes | no	no
birthday	<username>	Specifies the user's birthday.	<YYYYMMDD>	none
country	<username>	Specifies the user's country.	<string>	none
homepage	<username>	Specifies the user's homepage.	<string>	none
avatar	<username>	Specifies a 96x96 JPEG image to be displayed as the user's avatar picture. This may also be referred to as a buddy icon.	<system_path>	none
gender	<username>	Specifies the user's gender.	1 (male) | 2 (female)	none

There are some settings that will not be modified by issuing a reload command on the Asterisk CLI. The settings which will not be modified are *engine_directory*, *disable_tcpauto*, *disable_udp*, and *rtp_address*. The chan_skype.so Asterisk channel module must be fully unloaded and loaded again in order to change these values. This will occur anytime that Asterisk is restarted.

3.2 Dial Technology

The use of Skype for Asterisk channels is similar to other Asterisk channel drivers. The dial plan technology type provided by Skype for Asterisk is simply referred to as *Skype*. The following sections describe how to make outgoing calls and receive incoming calls using Skype for Asterisk.

3.2.1 Outgoing Calls

When calls are placed on the Skype community, they are placed to their destination by a Skype user associated with the Asterisk server. Skype for Asterisk must select one of the defined Skype users to be the originator of a call. For that purpose, the *default_user* option in the *chan_skype.conf* file can be set to control which user is the default originator of a call. Additionally, on a call-by-call basis, the originator of a call can be defined by prefixing the destination Skype user or SkypeOut number with the name of the originator's Skype user.

The syntax for making an outgoing call using Skype for Asterisk is as follows:

```
Dial(Skype/[<originator>@]<destination>)
```

The *destination* is mandatory and can be defined as a Skype user or a SkypeOut number. The *originator* is optional and can be defined as a Skype user associated with the Asterisk server. Both of the examples provided below would result in the *james_bond* Skype user placing the call to the destination.

```
exten =>  ...,1,Dial(Skype/james_bond@austin_powers)

exten =>  ...,1,Dial(Skype/james_bond@+12564286000)
```

The examples provided below show how to make an outgoing call by specifying only a destination Skype user or SkypeOut number. In these cases, the user specified in the *default_user* option in the *chan_skype.conf* file will be defined as the originator of the call.

```
exten => ...,1,Dial(Skype/austin_powers)

exten => ...,1,Dial(Skype/+12564286000)
```

3.2.2 Incoming Calls

Specified on a per-user basis, Skype for Asterisk can direct incoming calls to any desired dial plan context. Optionally, incoming calls can be directed to a specific extension within a context. The default configuration will use the name of the destination Skype user as the target extension.

If the *james_bond* Skype user is configured with *context=demo* in the *chan_skype.conf* file, then placing the following entries in the *extensions.conf* file will handle incoming calls for that user.

```
[demo]
exten => james_bond,1,NoOp(Incoming Skype Call!)
exten => james_bond,n,Dial(SIP/shoe-phone)
```

Skype users can be mapped to numeric extensions by specifying the *exten* option for that user in the *chan_skype.conf* file. If the *james_bond* Skype user is configured with *context=demo* and *exten=007* in the *chan_skype.conf* file, then placing the following entries in the *extensions.conf* file will handle incoming calls for that user.

```
[demo]
exten => 007,1,NoOp(Incoming Skype Call!)
exten => 007,n,Dial(SIP/shoe-phone)
```

3.2.2.1 Transfers

The *SkypeTransfer* application can be used to transfer an incoming Skype call and set a call topic via the dial plan.

The syntax for transferring an incoming Skype call and setting a call topic is as follows:

```
SkypeTransfer(<destination>[,<topic>])
```

The *destination* is mandatory and can be defined as a Skype user or a SkypeOut number. The *topic* is optional and can be defined as a text string. The example provided below would result in transferring an incoming Skype call to *austin_powers* with the call topic set as “Secret Call”.

```
[demo]
exten => james_bond,1,Answer
exten => james_bond,n,SkypeTransfer(austin_powers,Secret Call)
```

Note: Not all Skype clients display the call topic for non-group calls.

The *Transfer* application can alternatively be used to transfer an incoming Skype call via the dial plan when a call topic does not need to be set.

```
[demo]
exten => james_bond,1,Answer
exten => james_bond,n,Transfer(Skype/austin_powers)
```

3.3 Functions

The *CHANNEL* and *CALLERID* dial plan functions may be used to retrieve Skype values from a call that originates on a Skype channel. In addition, Skype for Asterisk provides a few native dial plan functions that can be used to set and retrieve values on the Skype community regardless of the type of channel that originated the call. The following sections describe these functions.

> **Note:** It is important to understand the meaning of the values in the *type* column of the function description tables provided in this section. Function options with a type of *RW* allow Read and Write access. Function options with a type of *RO* allow Read Only access.

3.3.1 Channel Function

Depending on privacy policies, multiple user details may be available about a caller's Skype user when connected to an Asterisk channel. These details can be retrieved by using the *CHANNEL* function in the dial plan. This function's syntax is as follows:

```
CHANNEL(<property>)
```

The Skype-related options available to the *CHANNEL* dial plan function are listed in the table below.

Property	Type	Description	Values
skype_language	RO	Reads a space-separated list of language identifiers for the call	<string> [<string> ...]
skype_topic	RO	Reads the call topic	<string>
skype_token	RO	Reads the access call token	<string>
skype_about	RO	Reads the caller's *about* profile	<string>
skype_birthday	RO	Reads the caller's birthday	<YYYYMMDD>
skype_gender	RO	Reads the caller's gender	1 (male) | 2 (female)
skype_homepage	RO	Reads the caller's home page	<string>
skype_homephone	RO	Reads the caller's home phone	<string>
skype_officephone	RO	Reads the caller's office phone	<string>
skype_mobilephone	RO	Reads the caller's mobile phone	<string>
skype_city	RO	Reads the caller's city	<string>
skype_province	RO	Reads the caller's province	<string>
skype_country	RO	Reads the caller's country	<string>

Below are descriptions of options that may not be intuitive.

- **skype_topic** – This option will retrieve the call topic. The call topic is a user-provided string that can identify the *topic* of the call. This commonly includes a URL with query parameters that can be used to dial a Skype user with a particular topic set.

- **skype_token** – This option will set or retrieve the access call token. If specified, Skype users who know the access call token can "call in" to the call.

The example provided below shows how to set the channel's language to the language that a caller's Skype user prefers. The language setting is read by prompt playback, the voicemail application, and many other Asterisk applications.

```
exten => ...,1,Set(CHANNEL(language)=${CHANNEL(skype_language)})
```

3.3.2 CallerID Function

The *CALLERID* function may be used to retrieve details about a caller's Skype user when connected to an Asterisk channel. These details can be retrieved by using the *CALLERID* function in the dial plan. This function's syntax is as follows:

```
CALLERID(<item>)
```

The Skype-related options available to the *CALLERID* dial plan function are listed in the table below.

Property	Type	Description	Values
name	RO	Reads the caller's full name if available.	<string>
num	RO	Reads the caller's PSTN number for a PSTN-based call, or the Skype account name for a Skype-to-Skype call.	<string>

The example provided below shows how to retrieve a caller's Skype account name for a Skype-to-Skype call.

```
exten => ...,1,NoOp(Caller's Skype account name is ${CALLERID(num)})
```

3.3.3 Skype Call Property Function

Since an outbound Skype call may originate from a non-Skype channel, the *CHANNEL* and *CALLERID* dial plan functions cannot always be used to retrieve Skype call properties. The *SKYPE_CALL_PROPERTY* function provides an interface to Skype in the Asterisk dial plan. It can be used to set and retrieve call properties, and to retrieve caller properties. This function's syntax is as follows:

```
SKYPE_CALL_PROPERTY(<property>)
```

The options available to the *SKYPE_CALL_PROPERTY* dial plan function are listed in the table below.

Property	Type	Description	Values
topic	RW	Specifies the call topic	<string>
token	RW	Specifies the access call token	<string>
forwarded_by	RO	Reads the Skype identity that forwarded the incoming call	<string>
target_identity	RO	Reads the SkypeIn number that was dialed on an incoming call	<string>
skypename	RO	Reads the caller's Skype account name	<string>
fullname	RO	Reads the caller's full name	<string>
country	RO	Reads the caller's country	<string>
province	RO	Reads the caller's province	<string>
city	RO	Reads the caller's city	<string>
phone_home	RO	Reads the caller's home phone	<string>
phone_office	RO	Reads the caller's office phone	<string>
phone_mobile	RO	Reads the caller's mobile phone	<string>
email	RO	Reads the caller's email address	<string>
homepage	RO	Reads the caller's homepage	<string>
about	RO	Reads the caller's *about* profile	<string>
birthday	RO	Reads the caller's birthday	<YYYYMMDD>
gender	RO	Reads the caller's gender	1 (male) | 2 (female)

Note: The value of *SKYPE_CALL_PROPERTY* function options that are *RW*

(Read/Write) can also be defined using the *setvar* parameter in the *chan_skype.conf* file.

Below are descriptions of options that may not be intuitive.

- **topic** – This option will set or retrieve the call topic. The call topic is a user-provided string that can identify the *topic* of the call. This commonly includes a URL with query parameters that can be used to dial a Skype user with a particular topic set.
- **token** – This option will set or retrieve the access call token. If specified, Skype users who know the access call token can “call in” to the call.
- **about** – This option will read the caller's *about* profile. Many Skype users include a short description of themselves in their *about* profile.

The example provided below shows how to set the *topic* call property.

```
exten => ...,1,Set(SKYPE_CALL_PROPERTY(topic)=Secret Plans)
```

The example provided below shows how to retrieve the *topic* call property.

```
exten => ...,1,NoOp(Topic is ${SKYPE_CALL_PROPERTY(topic)})
```

3.3.4 Skype Account Property Function

Since an outbound Skype call may originate from a non-Skype channel, the *CHANNEL* dial plan function cannot always be used to retrieve Skype call properties. The *SKYPE_ACCOUNT_PROPERTY* function provides an interface to Skype in the Asterisk dial plan. Skype account properties are stored on the Skype community and include information such as account availability, birthday, and geographical information. These settings can be set and retrieved using the *SKYPE_ACCOUNT_PROPERTY* dial plan function. This function's syntax is as follows:

```
SKYPE_ACCOUNT_PROPERTY(<account>,<property>)
```

The options available to the *SKYPE_ACCOUNT_PROPERTY* dial plan function are listed in the table below.

Property	Type	Description	Values
skypeout_currency	RO	Reads the Skype Credit balance currency type (USD, EUR, etc.).	<string>
skypeout_balance	RO	Reads the Skype Credit balance in cents. For example, a value of 4500 in EUR means 45.00 EUR.	<int>
fullname	RW	Specifies the Skype user's full name	<string>
country	RW	Specifies the Skype user's country	<string>
province	RW	Specifies the Skype user's province	<string>
city	RW	Specifies the Skype user's city	<string>
phone_home	RW	Specifies the Skype user's home phone	<string>
phone_office	RW	Specifies the Skype user's office phone	<string>
phone_mobile	RW	Specifies the Skype user's mobile phone	<string>
email	RW	Specifies the Skype user's email address	<string>
homepage	RW	Specifies the Skype user's homepage	<string>
about	RW	Specifies the Skype user's *about* profile	<string>
birthday	RW	Specifies the Skype user's birthday	<YYYYMMDD>
gender	RW	Specifies the Skype user's gender	1 (male) | 2 (female)
status	RW	Specifies the Skype user's status	Logged Out | Logged Out and Password Saved | Connecting to P2P Network | Connecting to Server | Logging In | Initializing | Logged In | Logging Out
mood_text	RW	Specifies the Skype user's “mood message”.	<string>
availability	RW	Specifies the Skype user's availability	Unknown | Offline | Online | Away | Not Available | Do Not Disturb | Invisible | Skype Me | Connecting

Note: The value of *SKYPE_ACCOUNT_PROPERTY* function options that are *RW* (Read/Write) can also be defined using the *setvar* parameter in the *chan_skype.conf* file.

Below is a description of options that may not be intuitive.

- **about** – This option will set or retrieve the Skype user's *about* profile. Many Skype users include a short description of themselves in their *about* profile.
- **mood_text** - Mood messages are simple little messages that tell your friends the mood you are in, a witty comment, quote, a web link or any random piece of information you'd like everyone to see.

The example provided below shows how to set the *gender* account property to male.

```
exten => ...,1,Set(SKYPE_ACCOUNT_PROPERTY(gender)=1)
```

The example provided below shows how to retrieve the *fullname* account property.

```
exten => ...,1,NoOp(Aston's full name is $
{SKYPE_ACCOUNT_PROPERTY(aston,fullname)})
```

3.3.5 Skype Buddy Functions

Skype for Asterisk provides a way to retrieve the buddy list and status of all buddies for a Skype user. This is accomplished by using the *SKYPE_BUDDIES* and *SKYPE_BUDDY_FETCH* dial plan functions.

The *SKYPE_BUDDIES* function will return an *id* to pass to the *SKYPE_BUDDY_FETCH* function to enumerate the buddies. The *SKYPE_BUDDIES* function's syntax is as follows:

```
SKYPE_BUDDIES(<account>)
```

The *SKYPE_BUDDY_FETCH* function will retrieve the next buddy, including status, from the buddy list *id* retrieved by the *SKYPE_BUDDIES* function. This information is returned as a string in the format of '*<buddy name>,<buddy status>'* . This format is suitable for use with the *ARRAY* dial plan function. The *SKYPE_BUDDY_FETCH* function's syntax is as follows:

```
SKYPE_BUDDY_FETCH(<id>)
```

The example provided below shows how to retrieve the status of all buddies that are on the *james_bond* Skype user's buddy list.

```
exten => james_bond,1,Set(id=${SKYPE_BUDDIES(${EXTEN})})
exten => james_bond,n,Set(ARRAY(buddy,status)=${SKYPE_BUDDY_FETCH($
{id})})
exten => james_bond,n,While($[${buddy}])
exten => james_bond,n,NoOp(${buddy} is ${status})
exten => james_bond,n,Set(ARRAY(buddy,status)=${SKYPE_BUDDY_FETCH($
{id})})
exten => james_bond,n,EndWhile()
```

3.3.6 Skype Chat Function

Skype for Asterisk provides a way to send and receive chat messages with other Skype users. This is accomplished by using the *SKYPE_CHAT_RECEIVE* function and the *SkypeChatSend* application in the dial plan.

The *SKYPE_CHAT_RECEIVE* function will wait to receive a Skype chat message from a user. The *SKYPE_CHAT_RECEIVE* function's syntax is as follows:

```
SKYPE_CHAT_RECEIVE(<to>,<from>,<timeout>)
```

Syntax Description:

- **to** - The Skype account receiving the message
- **from** - The Skype account that will be sending the message
- **timeout** - The number of seconds to wait for the message

The *SKYPE_CHAT_RECEIVE* function will receive a Skype chat message sent from "*from*" and to *"to"*. If no matching message arrives before "*timeout*", the return value will be empty.

The example provided below shows how to receive a Skype chat message.

```
exten => s,1,Set(message=${SKYPE_CHAT_RECEIVE(touser,fromuser,30)})
exten => s,n,NoOp(Received message: ${message})
```

The *SkypeChatSend* application will send a Skype chat message to a user. The *SkypeChatSend* application's syntax is as follows:

```
SkypeChatSend(from,to,message)
```

Syntax Description:

- **from** - The Skype account sending the message
- **to** – The Skype account that will be receiving the message

- **message** – The message to send (must be UTF-8 encoded)

The *SkypeChatSend* application will send the value of *“message”* as a Skype chat message sent from “*from”* and to *“to”*.

The example provided below shows how to send a Skype chat message.

```
exten => s,1,SkypeChatSend(fromuser,touser,message)
```

Note: Messages should be sent in plain text. Newlines are not permissible.

3.4 Hints

Unlike other Asterisk channel drivers, Skype for Asterisk does not manage devices at all. The concept of *device state* as represented by other channel drivers do not apply to Skype users. However, Skype for Asterisk does provide a mechanism for subscribing to and being notified of changes in the presence state of Skype users. Most interactive Skype clients display the presence state of Skype users using a graphical representation, and make their presence state available to Asterisk to be used by dial plan hints. Skype for Asterisk provides a custom device state provider called *Skype* that can be used with hints.

The example provided below shows how the *james_bond* Skype user can subscribe to the presence state of the *austin_powers* Skype user.

```
exten => ...,hint,Skype:james_bond@austin_powers
```

It is important to note the use of *':'* instead of '/' to separate the device state provider name from the item being watched. Using *'/'* will make Asterisk treat the *austin_powers* Skype user as a device and look for channels open to that Skype user. That would cause Asterisk to always report the *austin_powers* Skype user as busy. Using *':'* instead will cause Asterisk to *trust* the Skype for Asterisk module to report back the state of the *austin_powers* Skype user and to not infer what the state of that Skype user might be from other sources.

Similar to placing outgoing calls, presence state requests must originate from a Skype user associated with the Asterisk server. The difference is that there is no default user setting. A Skype user must always be prefixed to the Skype user whose state will be subscribed.

There is another important point to consider. The Skype community does not allow a user (User A) to see another user's (User B) state unless User A has added User B to his or her buddy list and User B has authorized that addition. In the example shown above, this means that the *james_bond* Skype user would have to add the *austin_powers* Skype user to his buddy list, and the *austin_powers* Skype user would need to authorize that addition.

Since Skype for Asterisk will often be configured to use Skype user accounts that are never used with an interactive Skype client, the *chan_skype.conf* file has a configuration parameter to make adding buddies easier. The *buddy_autoadd* option can be set to *buddies*, *hints*, or '*buddies,hints*'. When setting this option to *hints*, only buddies that receive a presence information request, such as from a dial plan hint, will be added to the buddy list. Setting this option to *buddies* will automatically add all buddies that have been authorized to receive that user's presence information to the buddy list. Lastly, setting this option to *'buddies,hints'* will cause both of these to occur.

In some cases, you may have Skype users logged in via Skype for Asterisk, but you are not interested in presence updates for those users' Skype buddies. In that case, you can disable presence updates by setting the *buddy_presence* option in the *chan_skype.conf* file for the relevant users.

3.5 Manager Commands

The Skype for Asterisk modules allow various manager commands to be issued by manager sessions that have the *SYSTEM* class manager permission. The manager commands listed below are handled by the Skype for Asterisk modules and detailed in this section.

- Skype Account Property
- Skype Add Buddy
- Skype Remove Buddy
- Skype Buddies
- Skype Buddy
- Skype Chat Send

3.5.1 Skype Account Property

The *SkypeAccountProperty* manager command can be used to set account properties for a Skype for Asterisk user. An example *SkypeAccountProperty* request and response are provided below.

```
Action: SkypeAccountProperty
ActionID: abc123
User: james_bond
variable: fullname=James Bond
variable: availability=Skype Me

Response: Success
ActionID: abc123
Message: Properties set
```

3.5.2 Skype Add Buddy

The *SkypeAddBuddy* manager command can be used to add a buddy to a Skype for Asterisk user's buddy list. An example *SkypeAddBuddy* request is provided below.

```
Action: SkypeAddBuddy
ActionID: abc123
User: james_bond
Buddy: austin_powers
AuthMsg: Please authorize me!
```

The *AuthMsg* line is optional and can be used to send an authorization request message when adding a Skype buddy. It is also the message that is used for authentication when someone tries to add an SfA user as a buddy and a password is specified with the *accept.<password>* value for the *auth_policy* setting in *chan_skype.conf*.

3.5.3 Skype Remove Buddy

The *SkypeRemoveBuddy* manager command can be used to remove a buddy from a Skype for Asterisk user's buddy list. An example *SkypeRemoveBuddy* request is provided below.

```
Action: SkypeRemoveBuddy
ActionID: abc123
User: james_bond
Buddy: austin_powers
```

3.5.4 Skype Buddies

The *SkypeBuddies* manager command can be used to retrieve the buddy list of a Skype for Asterisk user, including the full name and status of each buddy. An example *SkypeBuddies* request and response are provided below.

```
Action: SkypeBuddies
User: james_bond
ActionID: abc123

Response: Success
ActionID: abc123
Message: Skype buddy status list will follow

Event: SkypeBuddyEntry
ActionID: abc123
Buddy: echo123
Status: Waiting for Authorization

Event: SkypeBuddyEntry
ActionID: abc123
Buddy: austin_powers
Fullname: Austin Powers
Status: Offline (Voicemail Enabled)

Event: SkypeBuddylistComplete
ListItems: 2
ActionID: abc123
```

3.5.5 Skype Buddy

The *SkypeBuddy* manager command can be used to retrieve information about a specific buddy. An example *SkypeBuddy* request and response are provided below.

```
Action: SkypeBuddy
User: james_bond
Buddy: austin_powers
ActionID: abc123

Response: Success
```

```
ActionID: abc123
availability: Online
fullname: Austin Powers
birthday: 19450328
phone_mobile: +15555551212
```

3.5.6 Skype Chat Send

The *SkypeChatSend* manager command can be used to send a Skype chat message to a user. Messages must be UTF-8 encoded. An example *SkypeChatSend* request and response are provided below.

```
Action: SkypeChatSend
User: james_bond
Skypename: austin_powers
Message: Sending test message from AMI to skype client
ActionID: 222

Event: SkypeChatMessage
Privilege: system,all
SequenceNumber: 11
File: chan_skype.c
Line: 864
Func: new_chat_message
To: austin_powers
From: james_bond
Message: U2VuZGluZyB0ZXN0IG1lc3NhZ2UgZnJvbSBBTUkgdG8gc2t5cGUgY2xpZW50
```

Note: Newlines are not permissible in AMI messages. Therefore, messages sent in plain text must not include any newlines. The response "*Message*" is base64 encoded.

It is possible to send a message that includes one or more newlines by encoding the message in base64. In addition, the Encoding header must be included when sending a message in base64. An example is provided below.

```
Action: SkypeChatSend
ActionID: abc123
User: james_bond
Skypename: austin_powers
Encoding: base64
Message: RG9uJ3QNClJlYWQNClRoaXM=
```

Chapter 4: Troubleshooting

This chapter provides various methods for obtaining the necessary information to troubleshoot most problems relating to Digium's Skype for Asterisk. Multiple resources are available to obtain more information about Asterisk and Digium products. These resources are listed on page 58.

4.1 Manager Events

The Skype for Asterisk modules will send various types of manager events to manager sessions that are capable of receiving *SYSTEM* class manager events. The manager events listed below are sent by the Skype for Asterisk modules and detailed in this section.

- Skype Account Status Events
- Skype Buddy Status Events
- Skype Chat Message Events

4.1.1 Skype Account Status Events

One Skype account status event is always sent when a Skype for Asterisk user logs in or out of the Skype community. An example Skype account status event is provided below.

```
Event: SkypeAccountStatus
Privilege: system,all
Username: james_bond
Status: Logged In
```

The possible *Status* values are:

- Logged In
- Logged Out

4.1.2 Skype Buddy Status Events

Status changes for contacts in a Skype for Asterisk user's contact list will trigger a manager event. Below is an example Skype buddy status message.

```
Event: SkypeBuddyStatus
Privilege: system,all
Buddy: Skype/james_bond@austin_powers
BuddyStatus: Offline
```

The possible *BuddyStatus* values are:

- Waiting for Authorization
- Blocked
- Blocked Skypeout
- Skypeout
- Offline
- Online
- Away
- Not Available
- Do Not Disturb
- Skype Me
- Offline (Voicemail Enabled)
- Offline (Call Forwarding Enabled)
- Unknown

4.1.3 Skype Chat Message Events

Any Skype Chat message received will generate an AMI event. An example Skype chat message event is provided below.

```
Event: SkypeChatMessage
Privilege: system, all
Username: jbond
From: austin_powers
Message: U2t5cGUgRm9yIEFzdGVyaXNrIHJvY2tzIQ==
```

Note: The *"Message"* field is base64 encoded since embedded newlines are not permissible in AMI messages.

4.2 Asterisk Command Line Interface (CLI)

The Asterisk CLI provides the operations in the list below:

- skype login user <username>
- skype logoff user <username>
- skype show buddies <username>
- skype set debug off [user] [<username>]
- skype set debug on [user] [<username>]
- skype show hostid
- skype show licenses
- skype show settings
- skype show users
- skype show user <username>
- skype set vedebug off <username>
- skype set vedebug on <username>
- skype show version

4.2.1 skype login user <username>

This CLI operation logs in the specified user on the Skype community.

4.2.2 skype logoff user <username>

This CLI operation logs off the specified user on the Skype community.

4.2.3 skype show buddies <username>

This CLI operation displays a list of buddies for the specified Skype for Asterisk user.

4.2.4 skype set debug off [user] [<username>]

This CLI operation disables global Skype debugging or user specific Skype debugging.

4.2.5 skype set debug on [user] [<username>]

This CLI operation enables global Skype debugging or user specific Skype debugging.

4.2.6 skype show hostid

This CLI operation displays the Skype for Asterisk Host-ID.

4.2.7 skype show licenses

This CLI operation displays the Skype for Asterisk licenses.

4.2.8 skype show settings

This CLI operation displays the global Skype for Asterisk configuration.

4.2.9 skype show users

This CLI operation displays a list of Skype for Asterisk users and their statuses.

4.2.10 skype show user <username>

This CLI operation displays the specified user's Skype for Asterisk configuration.

4.2.11 skype set vedebug off <username>

This CLI operation disables Voice Engine debugging for the specified Skype for Asterisk user.

4.2.12 skype set vedebug on <username>

This CLI operation enables Voice Engine debugging for the specified Skype for Asterisk user.

4.2.13 skype show version

This CLI operation displays the version of the Skype for Asterisk modules that are loaded.

4.3 Frequently Asked Questions

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

Is Skype for Asterisk available on an operating system other than Linux?

No, Skype for Asterisk is available for Linux only.

What branches of Open Source Asterisk are compatible with Skype for Asterisk?

Open Source Asterisk branches 1.4 (release 1.4.25 or newer), 1.6.2 (release 1.6.2.0 or newer), and 1.8 (release 1.8.0 or newer) are compatible.

Does Skype for Asterisk provide the same capabilities when used with Open Source Asterisk 1.4, 1.6.2, and 1.8?

Yes.

What configuration file(s) must be modified?

The */etc/asterisk/chan_skype.conf* file is a mandatory configuration file that will need to be modified to meet your specific needs.

Should I add a load line for res_skypeforasterisk and/or chan_skype 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_skypeforasterisk.so or chan_skype.so files. Asterisk will automatically load them using the autoload option. The autoload option is enabled by default.

Does issuing the reload command on the Asterisk CLI reload all of the Skype for Asterisk settings?

No. There are some settings that will not be modified by issuing a reload command on the

Why is there a process by the name of skypeforasterisk running when Asterisk is loaded?

The res_skypeforasterisk.so module contains a binary Skype engine called skyhost. Skype for Asterisk communicates with skyhost to make and manage connections to the Skype community. This engine automatically runs as a separate Linux process called skypeforasterisk once chan_skype.so is loaded.

Why is there an XML database file for Skype being stored under Asterisk's spool directory?

The Skype engine creates a small database of information for users that it logs in to the Skype community. This database is stored in a sub-directory called *skype* under Asterisk's spool directory. By default, Asterisk's spool directory is located at */var/spool/asterisk*. If your *asterisk.conf* specifies a different path for this directory using the *astspooldir* option, then Skype for Asterisk will use that directory instead.

I receive a warning or error from SELinux regarding one of the Skype for Asterisk modules when Asterisk starts. This prevents Skype for Asterisk from properly functioning. How do I resolve this?

There are two resolutions to this issue. The first involves disabling SELinux using the steps shown below.

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

If the use of SELinux is mandated by you or another authority within your organization, use the following command to give the res_skypeforasterisk.so module the proper execution privileges:

chcon -t texrel_shlib_t /usr/lib/asterisk/modules/res_skypeforasterisk.so

A symptom of this issue is a message similar to the following:

“cannot restore segment prot after reloc: Permission denied”

Asterisk did not cleanly shut down. Now Skype for Asterisk does not properly function when Asterisk starts. How do I resolve this?

If Asterisk is not shut down cleanly, the skypeforasterisk process may still be running on your system. If that is the case, follow the steps shown below.

1. Execute *'ps ax'* to determine the process ID (PID) of the *skypeforasterisk* process.
2. Execute *'kill -9 <PID>'*.
3. Restart Asterisk.

What defines a channel of Skype for Asterisk?

A single concurrent call on the Skype community

How many users can share a concurrent call?

Each user making a call will use a channel. For calls from one user to another user managed on the same Asterisk server, 2 concurrent calls will be used.

Can I use Skype for Asterisk on Switchvox systems?

Not at this time. This is planned for a future release of Switchvox.

Can I use Skype for Asterisk with AsteriskNOW?

Yes.

Will Skype for Asterisk run on other open source distributions of Asterisk such as TrixBox CE?

Yes. This will require manual configuration unless a 3rd-party GUI wrapper is created for those systems.

Will production systems be able to use Skype for Asterisk without reinstalling?

Yes.

Is there a monthly charge for using Skype for Asterisk?

No. There is a one time charge for each channel.

Is G.729 included with each channel of Skype for Asterisk?

Yes. The Digium G.729 software codec module (codec_g729a.so) will need to be installed in addition to Skype for Asterisk. Your Skype for Asterisk license key may be used to activate a G.729 channel at no additional cost. This requires version 3.0.0 or later of the Digium G.729 software codec module. This allows Skype for Asterisk users to use the G.729 codec for their Skype calls. This is commonly required for SkypeIn and SkypeOut calls. For more information regarding the Digium G.729 software codec module, please read the G.729 README that is available in the documentation section at http://www.digium.com/support.

The current version of chan_skype.so does not provide passthrough G.729 support like other Asterisk channel drivers. When a Skype call wants to use G.729, the codec_g729a.so module must be loaded, and G.729 licensed channels must be available. This will be improved in a future release of Skype for Asterisk. For this release, it is suggested to configure your users to only allow G.711 ulaw and/or alaw in the chan_skype.conf file unless you have G.729 licensed channels available.

What components are provided with Skype for Asterisk?

The Skype for Asterisk product consists of two Asterisk loadable modules:

- *res_skypeforasterisk.so* - This module contains the Skype engine, along with various libraries and other components required to talk to the Skype engine and manage user accounts, calls, presence, etc. This module is provided in a binary-only form.

- *chan_skype.so* - This module is the Asterisk channel driver that provides calling services to and from the Skype community, using the library services provided by res_skypeforasterisk.so.

Will Skype for Asterisk support any type of Skype user?

No. Only accounts created from the Business Control Panel will be usable with Skype for Asterisk. The administrator account for the Business Control Panel is a regular Skype account. Due to this fact, the administrator account will not be able to use Skype for Asterisk.

How can I access the Skype Business Control Panel?

Visit Skype's web site at *http://www.skype.com/business*.

Is there a cost to use the Skype Business Control Panel?

No. It is a web-based tool that is free to setup and use.

Can Skype for Asterisk and Skype For SIP coexist on the same Asterisk Server?

Yes.

How do I purchase Skype for Asterisk?

- End users: A Digium reseller
- Resellers: A Digium Distributor
- Distributors: Direct from Digium
- For those not serviced by a reseller: Digium direct at *https://www.digium.com/skype*.

Where can I find knowledge base articles for Skype for Asterisk?

Please visit the Skype for Asterisk category of the Digium Knowledge Base:
http://kb.digium.com/?CategoryID=273

How do I get support for Skype for Asterisk?

Skype for Asterisk comes with installation support for the first 90 days from purchase. If you need support, please contact Digium's support team at *http://www.digium.com/support*. For subscriptions covering Open Source Asterisk or Asterisk Business Edition, one incident can be used to support Skype for Asterisk with a current subscription.

What do I submit to Support when I'm having Skype problems?

Perform the following steps:

1. At the Asterisk CLI, type "skype set debug on".
2. At the Asterisk CLI, type "skype set vedebug on user <username>" for the user that is having the problem.
3. At the Asterisk CLI, type "core set verbose 6". Verbosity can be 6 or higher.
4. At the Asterisk CLI, type "skype show version".
5. Redirect a manager session (with *SYSTEM* class permissions) to a file.
6. Reproduce the issue.
7. Submit Asterisk CLI output and manager session output to Support.

Where can I 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 send an e-mail to sales@digium.com.

Appendix A: Glossary and Acronyms

ANSI *American National Standards Institute*

An organization that 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 (usually owned and operated by an ILEC).

CPE *customer premises equipment*

Terminal equipment that is connected to the telecommunications network and that 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 simultaneous directions.

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 that 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 that 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 that 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 that 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 B: DIGIUM END-USER PURCHASE AND LICENSE AGREEMENT

July 2009

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G.729 for Asterisk

FAX for Asterisk

HPEC for Asterisk

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FAX for Asterisk

G.729 for Asterisk

HPEC for Asterisk

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