TeleConnx

VERSION 1.0
TECHNICAL MANUAL

APRIL 2023

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TeleConnx

TeleConnx offers custom long distance telecom solutions and messaging capabilities tailored to meet the unique needs of businesses. With a dedicated team experienced in building both simple and complex Asterisk-based telephone systems, TeleConnx provides flexible, reliable, and affordable communication solutions. Their support contracts include a 10x5 schedule with a 30-minute response time guarantee, emphasizing their dedication to timely and efficient customer service. Founded in 2013 in Cripple Creek, Colorado, TeleConnx serves businesses worldwide with a commitment to providing unique telecom solutions.

Cripple Creek, Colorado

Cripple Creek, Colorado, 0204769

Phone: (877) 630-5100 | https://www.teleconnx.com/

TeleConnx

Dear Clients,

At TeleConnx, we are dedicated to providing customized solutions for your unique telecom needs. Our robust communication solutions allow for flexible, reliable, and high-quality calling at an affordable price. Whether you are a small business or a large enterprise, we have the expertise to design and support telecom capabilities tailored to your specific requirements.

In addition to our customized telecom solutions, we also provide messaging solutions that enable you to interact with your customers across multiple platforms. Our team has years of experience in building both simple and complex Asterisk-based telephone systems for clients in various industries, including healthcare, finance, and manufacturing.

We take pride in being there for you when you need us the most. With our support contracts, we offer a 30-minute response time guarantee and are available on a 10x5 schedule, an extra hour every day that our competitors do not offer. Our dedication to timely and efficient support ensures that your business can operate smoothly and without interruption.

At TeleConnx, we understand the importance of keeping up with the latest technologies and trends in the telecom industry. That's why we stay up-to-date with the latest advancements to offer cutting-edge solutions that can help your business stay ahead of the competition.

Since our founding in Cripple Creek, Colorado, in 2013, we have been committed to providing exceptional telecom solutions to businesses around the world. Our mission is to provide the highest quality of service and support to our clients and to build long-lasting relationships based on trust and reliability.

Thank you for considering TeleConnx as your telecom partner. We look forward to working with you to develop the ideal solution for your business needs.

Sincerely,

The TeleConnx Team

W

William Howell, CEO
TeleConnx Incorporated

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TERMINOLOGY

Asterisk: An open-source software platform for building communication applications, including VoIP (voice over IP) and PBX (private branch exchange) systems.

PBX: A private branch exchange is a telephone system within an enterprise that switches calls between internal users and connects them to the public switched telephone network (PSTN).

VoIP: Voice over IP is a technology that enables voice communication and multimedia sessions over the internet or other IP-based networks.

SIP: Session Initiation Protocol is a signaling protocol used for initiating, maintaining, and terminating real-time sessions that involve voice, video, messaging, and other communications applications.

RTP: Real-Time Transport Protocol is a network protocol used to transport multimedia data, including audio and video, over IP networks.

Codec: A codec is a device or software application used to encode or decode digital signals, including audio and video data.

API: An Application Programming Interface is a set of protocols, routines, and tools for building software applications that can access the features or data of another application, operating system, or service.

DNS: Domain Name System is a protocol that translates domain names into IP addresses, allowing users to access websites and other resources on the internet using human-readable names instead of numerical IP addresses.

QoS: Quality of Service is a set of network protocols and technologies used to ensure that data packets are delivered with a minimum level of latency, jitter, and packet loss, particularly for real-time communications applications such as VoIP.

Latency: The amount of time it takes for a packet of data to travel from one point to another on a network, typically measured in milliseconds (ms). Latency can affect the quality of real-time communications applications such as VoIP, video conferencing, and online gaming.

Jitter: Variations in the latency of packets as they travel across a network, which can cause disruptions or delays in real-time communications.

Bandwidth: The amount of data that can be transmitted over a network in a given amount of time, typically measured in bits per second (bps) or kilobits per second (Kbps). Bandwidth can affect the quality and speed of data transmission, including voice and video.

DNSSEC: Domain Name System Security Extensions is a protocol that provides an additional layer of security for DNS lookups by adding digital signatures to DNS records, helping to prevent DNS spoofing and other attacks.

INTRODUCTION

Welcome to the technical manual for TeleConnx's Asterisk Speech-to-Text Integration for Deepgram, the world's most powerful speech-to-text model. This comprehensive guide will help you understand how to utilize our integration on any Asterisk PBX system. This manual is designed for both technical and non-technical personnel who are responsible for the operation and maintenance of Asterisk-based systems.

This manual provides an overview of our Deepgram Integration for Asterisk; installation procedures, operating instructions and troubleshooting guidelines. It is intended to be used as a reference guide for those who are responsible for the day-to-day operation of Asterisk systems, as well as for those who are involved in the design of custom dialplans.

The manual is divided into several sections, each covering a different aspect of our systems. These sections include:

Terminology: A glossary of commonly used technical terms and acronyms that are specific to our systems.

System Overview: A detailed description of the architecture and components of our telecommunication systems, including hardware and software components.

Installation: Step-by-step procedures for installing our systems in various environments, including hardware and software requirements, cabling, and configuration.

Configuration: Instructions for configuring our systems to meet specific client requirements, including call routing, messaging protocols, and other customization options.

Operation: Procedures for operating our systems, including starting and stopping services, monitoring performance, and managing user accounts.

Troubleshooting: A comprehensive guide to diagnosing and resolving common issues that may arise during the operation of our systems.

We encourage you to use this manual as a reference guide to help you become familiar with our systems and to ensure their reliable operation. If you have any questions or need further assistance, please don't hesitate to contact our technical support team for assistance.

HOW TO BUY

Thank you for considering TeleConnx's Asterisk Speech-to-Text Integration for Deepgram – the fastest, most robust and reliable speech-to-text solution on the market. Our team is dedicated to providing you with the fastest and most reliable telephone solutions at an affordable price.

Buy Online: **www.teleconnx.com/dg-stt/buy**

You can purchase our integration from this website link.

Lightning Fast: Highly multi-threaded to operate in high-volume call centers handling tens of thousands of calls per hour, providing extremely low-latency interactions. Compile-time optimizations to ensure fastest performance on various platforms, including ARM architectures.

Rock Stable: Designed to run 24/7 and not require restarts; from licensing changes, to endpoint changes, to configurations – can all be updated without a restart. Written on top of our core libraries that have been proven over 15 years in call center environments

Tiny Footprint: Designed to run on on any size system; utilizing just 28MB of RAM. We know that telephone infrastructure comes in a variety of sizes, and we have built our software to have minimal impact on any type of server; even the smallest virtual ones.

Telephone Audio: Audio support for real-time transcribing of common telephony codecs like uLaw, aLaw, GSM and G.722. Optimized to provide the best quality transcriptions of 8kHz telephone audio.

Flexible Calling: Change your transcription settings in the dial plan, like language, api key, and tagging right in your call flow; so, whether you need to use a client's key for a specific call, or handle transcription in another language, you can manage that right in your dial plan.

Visit our website, **www.teleconnx.com** to get in touch with us, should you need more information or other payment options.

INSTRUCTIONS

Login to your phone system via **SSH**, using **PuTTY** or **Terminal**.

From your PBX login prompt, if you didn't login directly as root, escalate your privileges to root with the following:

```
$ sudo su -
```

Now run our installation script process as follows:

```
# wget -O /tmp/dg-stt-kickstart.sh http://cdn.nxev.com/tcx/dg-sttkickstart.sh &&
sh /tmp/dg-stt-kickstart.sh
```

Or, if you don't have wget installed, you can use curl as follows:

```
# curl http://cdn.nxev.com/tcx/dg-stt-kickstart.sh > /tmp/dg-sttkickstart.sh &&
sh /tmp/dg-stt-kickstart.sh
```

Confirm that Asterisk has the AEAP modules installed and running, as our integration requires it:

```
# asterisk -x "module show like aeap"
```

We offer a free 30-day trial, so you can try the unlimited channel version of our software right away, without having to purchase it.

If the modules are installed and running, you should see the following:

```
Module                                    Description
Use Count Status                          Support Level
res_aeap.so                               Asterisk External Application
Protocol M 1                              Running core
res_speech_aeap.so                        Asterisk External Application Speech
Eng 0                                     Running core
2 modules loaded
```

Our integration is installed as a service, and can be managed with systemd. You can confirm that the service is running with:

```
# systemctl status tcx-dg-stt
```

INSTRUCTIONS

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The configuration file for our service can be found at **/etc/tcx-dgstt.conf** and you'll find all possible configuration settings there, with most commented-out. The first required change will be to update the "key" and set that to your Deepgram API Key you created within your Deepgram portal.

The more important Deepgram settings are:

key: Your Deepgram API key **required** to start transcribing.
tier: Level of model you would like to use in your request.
model: AI model used to process submitted audio.
language: The language of the audio to transcribe.
keywords: Keywords to which the model should pay particular attention.
tag: Tag to associate with the request.

You can find all the information on Deepgram-related configuration settings at :

https://developers.deepgram.com/api-reference/transcription/#transcribe-live-streaming-audio.

Please note that any changes made to our configuration file become "live" immediately. If a change breaks the integration, we will do our best to fallback on the last known working configuration; however, you should be aware that unsupported combinations of settings could cause the speech-to-text to stop functioning.

We install a sample dialplan in **/etc/asterisk/extensions_custom.conf** called deepgram-demo which you can utilize for testing. This dialplan will loop, saying "hello" each time, and then wait for speech, which is then output into the logs. The call will exit immediately when "goodbye" is heard in the audio stream.

TROUBLESHOOTING

The most common failures are related to **an incorrect** Deepgram API key or an **invalid** combination of configuration options, such as using the "phonecall" model with languages other than English that do not support that model. Additionally, if a call is using an unsupported codec, the connection will fail and SpeechStart will not succeed – the logs will show an unsupported codec.

CLOSING

Thank you for taking the time to read through our technical manual for TeleConnx's Asterisk Speech-to-Text Integration for Deepgram. We hope that you found the information contained within this document to be helpful and informative. Our goal is to provide you with the tools and knowledge you need to effectively utilize our products and services, and it is our sincere hope that this manual helped you in that regard.

If you have any questions or concerns about the material presented here, please don't hesitate to reach out to our support team. We're always happy to help and can be reached by phone, email, or through our website.

In addition to the resources provided in this manual, we also offer a range of training and support services to help you get the most out of our solutions. Whether you're a first-time user or an experienced professional, we're here to help you succeed.

Once again, thank you for choosing TeleConnx for all your telephone needs. We appreciate your business and look forward to serving you in the future.