2600Hz

PBX Connector

App Overview

SIP Trunks for Legacy PBXes

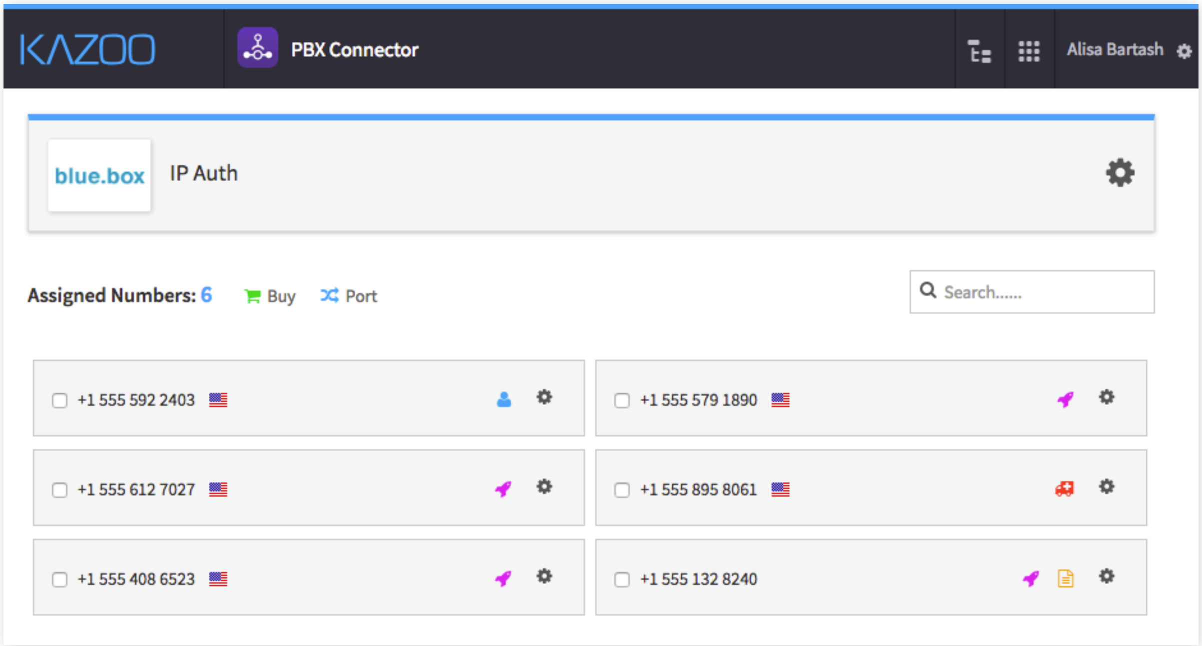
PBX Connector

Overview

PBX Connector establishes a connection between a PBX phone and KAZOO, allowing you to sell phone service to companies using legacy PBX systems. With just a few clicks, you can authenticate and configure any PBX and increase revenue by taking on customers not looking to abandon on-premise equipment, but who are still looking to lower costs and gain feature benefits of a VoIP service.

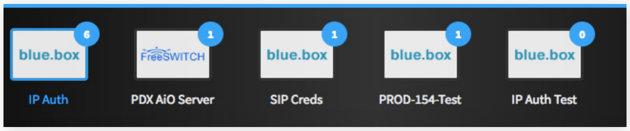
With the PBX Connector app, you can:

- Migrate customers to a cloud solution without changing their entire system at once
- Setup any SIP compatible system to utilize VoIP
- Onboard customers with no disruption to their service – customers keep their old PBX phones and features
- Support inbound DIDs/DNIS
- Connect equipment behind any IP address, such as door phones, overhead paging units, etc.
- Support static IP or password login
- Integrate PBXes seamlessly



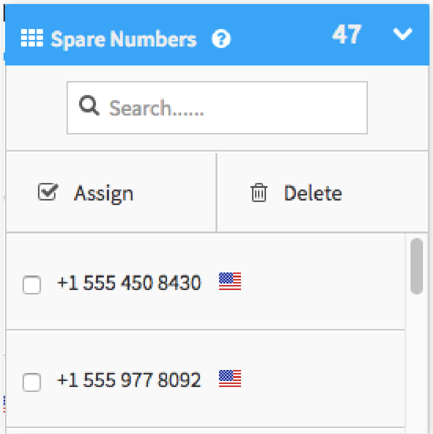
PBX Connector

Highlights



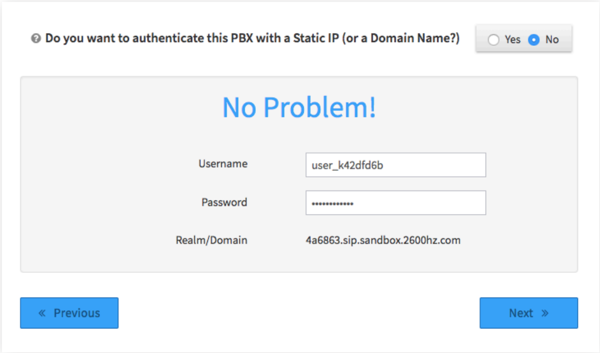
Connect and Support PBXes

Connect and support multiple PBXes within a single account, and assign each PBX its own unique number.



Integrated Number Management

Search, buy, port, assign, and manage numbers in one place, all without leaving the PBX Connector app. View numbers already assigned and update settings including caller-ID, e911, prepend, and failover.

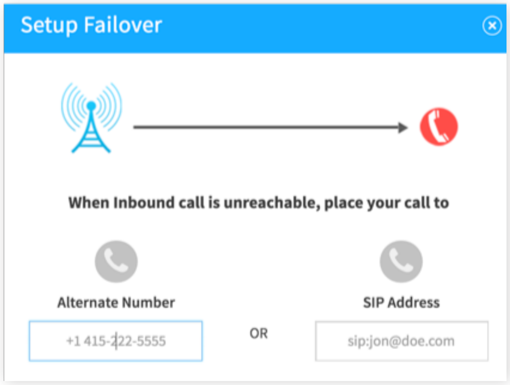


Fast Device Authentication

Choose to authenticate devices with a static IP or domain name, or with a username, password and realm/domain.

Failover and Alternate Routing

Customize what happens on a per-number basis when the customer's PBX or internet aren't working. A valuable feature for any organization, failover allows you to route calls to an alternate service in the event of a failure.



PBX Connector

Technical Specifications

Features

- Multiple PBXes in a single account
- Point-to-Point Audio (Proxy Mode)
- Caller ID / Caller ID Name Support
- Failover to SIP URI or alternate off-net number
- IP based authentication
- Username/password authentication

Request URI / INVITE Format

- E.164
- 1 NPA NXX
- NPA NXX
- SIP_Username

DTMF

- SIP INFO
- RFC2833

Fax

- T.38 (best effort)

Audio

- PCMU / G711u
- PCMA / G711a
- G729
- G722 / G722.1
- OPUS
- Speex

Video

- H.263
- H.264
- VP8 / VP9

Phone Number Services

- Buy local, toll-free, and vanity numbers
- Port in numbers
- Caller-ID Outbound (LIDB updates)
- Caller-ID Inbound (CNAM lookups)
- E911 Provisioning / Management

Compatible PBX Manufacturers*

- blue.box
- FreeSWITCH
- 3com
- Allworx
- Altigen Communications
- Asterisk
- Avaya
- Cisco
- Digium
- epygi
- FreePBX
- Mitel
- ObjectWorld
- pingtel
- Response Point
- Samsung
- ShoreTel (IP based authentication only)
- Sutus
- TalkSwitch
- Taridium

* Supports most systems which adhere to general SIP signaling standards and utilize listed codecs

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