A New Day: Introducing Incredible PBX 2021 for Ubuntu 20.04

🌐 nerdvittles.com/a-new-day-introducing-incredible-pbx-2021-for-ubuntu-20-04

We've completed the rollout of Incredible PBX 2021 for Debian 10 and, thanks to the terrific work of @jaminmc, today we're pleased to introduce Incredible PBX 2021 for Ubuntu 20.04 LTS with its ten-year life cycle. For the next few weeks, you probably should consider this beta, non-production code because we still are sorting out a few wrinkles. But everything looks good, and we're comfortable releasing this as Ubuntu has its roots in Debian.

In addition to @jaminmc, we also want to offer our special thanks to the many talented individuals on the VoIP-Info.org Forum who work through the growing pains of these new releases to bring you open source products at zero cost. Come join the party!

If you're using on-premise hardware, begin by downloading the netinst ISO image of Ubuntu 20.04 for amd64. Follow our previous tutorials for tips on installation with VirtualBox or VMware ESXi. If you'd prefer to experiment in the cloud for about a penny an hour, open an account at Vultr or Digital Ocean using our referral links that support the Nerd Vittles project. You'll also get some free credit to try out the service. Then create a new $5/month Ubuntu 20.04, 64-bit instance in your favorite city. Want some cheaper KVM cloud alternatives? Visit the Incredible PBX Wiki.

If your PBX is sitting behind a NAT-based router, you'll need to redirect incoming UDP 5060-5061 and UDP 10000-20000 traffic to the private IP address of your server. This is required for all of the SIP providers included in the Incredible PBX 2021 build. Otherwise, all inbound calls will fail.

Installing Incredible PBX 2021 on Ubuntu 20.04 Server

Once your Ubuntu 20.04 platform is up and running, login as root using SSH or Putty. If you wish to use the default Asterisk 18 platform, issue the commands below to kick off the install. If you'd prefer to stick with Asterisk 16 for the time being and perhaps upgrade to Asterisk 18 later with the included upgrade script, then edit the script once you download it and change 18 to 16 in line 34 of the script before executing it.

```
wget http://incrediblepbx.com/IncrediblePBX2021-U.sh
chmod +x IncrediblePBX2021-U.sh
./IncrediblePBX2021-U.sh
```

At about 5 minutes into the install procedure, you'll be prompted for your telephone country code. It's 1 for Canada and the United States. Once the installation completes, reboot and you should be good to go.

```
Incredible PBX 2021.01U for Ubuntu 20.04

Asterisk: UP    Apache/5.6: UP      MySQL: UP
SendMail: UP     IPtables: UP        SSH: UP
LAN port: UP     Fail2Ban: UP     Webmin: UP
UCP Dmon: UP     PortKnock: UP     NR VPN: UP
FaxGetty: UP     IAX Modem: UP    HylaFax: UP

RAM:111896   Ubuntu Focal Fossa   Disk:14G

Asterisk 18.2.0    Incredible GUI 15.0.12.40

LAN: 140.82.30.149

Public Info: 140.82.30.149

System Time: Mon 25 Jan 2021 03:25:59 PM UTC


               <  OK  >
```
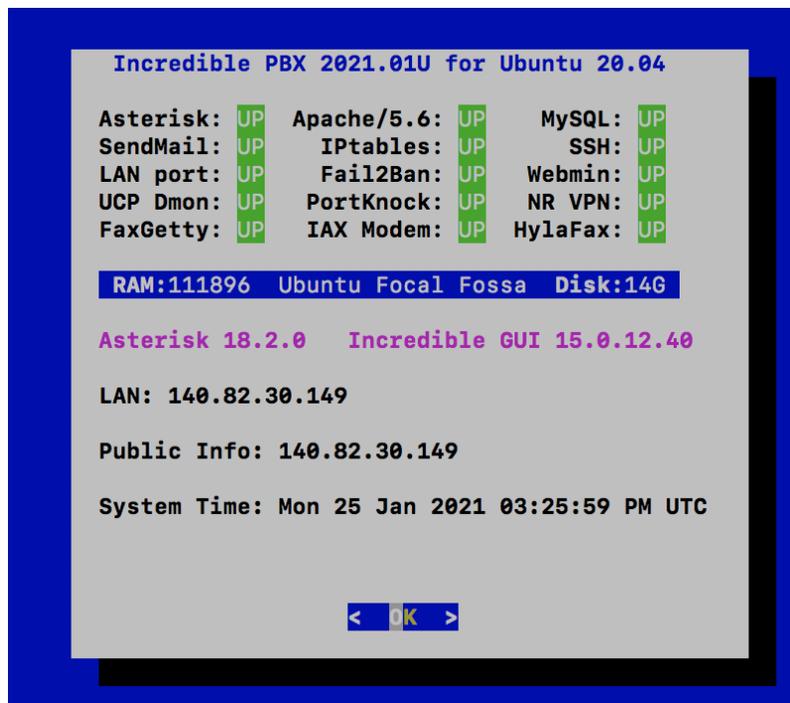
Next Steps with Incredible PBX 2021

Before you can manage your PBX through a web browser, you first will need to set the admin passwords for FreePBX, Apache web apps such as Reminders and AsteriDex, and AvantFax (if you are using Incredible Fax). These all can be set by logging into your server as root and issuing the following commands: *admin-pw-change*, *apache-pw-change*, and *avantfax-pw-change*. The AvantFax password can also be reset with a browser by first logging in as admin with a password of password.

Outbound mail functionality needs to be working so that you can receive voicemail messages and faxes by email. To prevent SPAM, most ISPs and ITSPs block messages from downstream mail servers. That would be you. The easiest way to resolve this is to configure

SendMail using Gmail as an SMTP Smarthost. You obviously need a Gmail account to implement this and, if you have turned on two-factor authentication for your Gmail account, you also will need to <u>obtain an App password</u> for your Gmail account, and use that in lieu of your regular Gmail password when configuring SendMail. With your Gmail username and password in hand, log into your server as root and run: **/root/enable-gmail-smarthost-for-sendmail**.

If your Incredible PBX 2021 is hosted with a cloud provider, be advised that many providers do not include a swap file as part of their offering. FreePBX requires a swap file. To add one, issue this command after logging into your server as root: **/root/create-swapfile-DO**.

To correctly set the time on your PBX, run: **/root/timezone-setup**.

By default, the voicemail password for each of the configured extensions (701-705) is set to the extension number. This means the user will be prompted to set a voicemail password on the first login to voicemail for each extension. A phone must be registered to the actual extension to access its voicemail account. For example, once a phone is registered to extension 701, the voicemail setup can be accessed by dialing *98701.

If you want to send and receive faxes with Incredible PBX 2021, run the following script while logged into your server as root: **/root/incrediblefax2020-debian10.sh**. When prompted, accept all the defaults. Once the HylaFax and AvantFax components are installed, reboot your server. To send faxes, click on the AvantFax tab in the FreePBX GUI and enter your login credentials (default: admin/password). To receive incoming faxes once you have configured a trunk and DID for your PBX, login to the FreePBX GUI as admin. Navigate to *Connectivity -> Inbound Routes*. For each of your DIDs on which you wish to receive faxes, select the inbound route and click the *Fax* tab. Review the Default Inbound Route Fax settings for proper setup.

NOTE: If you get a database error when you attempt to access AvantFax from a web browser, log into your server as root and reinstall the AvantFax database: `./avantfax-db-replace` .

Overview of the Initial Asterisk Setup Process

For those new to PBXs, here's a two paragraph summary of how Voice over IP (VoIP) works. Phones connected to your PBX are registered with Extensions so that they can make and receive calls. When a PBX user picks up a phone and dials a number, an Outbound Route tells the PBX which Trunk to use to place the call based upon established dialing rules. Unless the dialed number is a local extension, a Trunk registered with some service provider accepts the call, and the PBX sends the call to that provider. The provider then routes the call to its destination where the recipient's phone rings to announce the incoming call. When the recipient picks up the phone, the conversation begins.

Looking at things from the other end, when a caller somewhere in the world wishes to reach you, the caller picks up a telephone and dials a number known as a DID that is assigned to you by a provider with whom you have established service. When the provider receives the call to your DID, it routes the call to your PBX based upon destination information you established with the provider. Your PBX receives the call with information identifying the DID of the call as well as the CallerID name and number of the caller. An Inbound Route on your PBX then determines where to send the call based upon that DID and CallerID information. Typically, a call is routed to an Extension, a group of Extensions known as a Ring Group, or an IVR or AutoAttendant giving the caller choices on routing the call to the desired destination. Once the call is routed to an Extension, the PBX rings the phone registered to that Extension. When you pick up the phone, the conversation begins.

Configuring Trunks with Incredible PBX GUI

Perhaps the most difficult component to configure in the PBX is the Trunk. Almost every provider has a different way of doing things. We've taken some of the torture out of the exercise by providing a script which will configure settings for dozens of providers in seconds. Once installed, all you need to do is edit the desired Trunk (**Connectivity:Trunks**), change the **Disable Trunk** entry to *No*, and insert your credentials in both the **PEER Details** and **Registration** string of the **SIP Settings** *Outgoing* and *Incoming* tabs. Skyetel is enabled by default and needs no setup on the PBX side.

Configuring Skyetel for Incredible PBX 2021

If you've decided to go with Skyetel, here's the drill. Sign up for Skyetel service and take advantage of the Nerd Vittles specials. First, complete the Prequalification Form here. You then will be provided a link to the Skyetel site to complete your registration. Once you have registered on the Skyetel site and your account has been activated, open a support ticket and request the $10 credit for your account by referencing the Nerd Vittles special offer. Once you are satisfied with the service, fund your account as desired, and Skyetel will match your deposit of up to $250 simply by opening another ticket. That gets you up to $500 of half-price calling. Credit is limited to one per person/company/address/location.

Skyetel does not use SIP registrations to make connections to your PBX. Instead, Skyetel utilizes **Endpoint Groups** to identify which servers can communicate with the Skyetel service. An Endpoint Group consists of a Name, an IP address, a UDP or TCP port for the connection, and a numerical Priority for the group. For incoming calls destined to your PBX, DIDs are associated with an Endpoint Group to route the calls to your PBX. For outgoing calls from your PBX, a matching Endpoint Group is required to authorize outbound calls through the Skyetel network. Thus, the first step in configuring the Skyetel side for use with your PBX is to set up an Endpoint Group. Here's a typical setup for Incredible PBX 2021:

- **Name:** MyPBX

- **Priority:** 1
- **IP Address:** PBX-Public-IP-Address
- **Port:** 5060
- **Protocol:** UDP
- **Description:** my.incrediblepbx.com

To receive incoming PSTN calls, you'll need at least one DID. On the Skyetel site, you acquire DIDs under the **Phone Numbers** tab. You have the option of **Porting in Existing Numbers** (free for the first 60 days after you sign up for service) or purchasing new ones under the **Buy Phone Numbers** menu option.

Once you have acquired one or more DIDs, navigate to the **Local Numbers** or **Toll Free Numbers** tab and specify the desired **SIP Format** and **Endpoint Group** for each DID. Add **SMS/MMS** and **E911** support, if desired. **Call Forwarding** and **Failover** are also supported. That completes the VoIP setup on the Skyetel side. System Status is always available here.

Configuring VoIP.ms for Incredible PBX 2021

To sign up for VoIP.ms service, may we suggest you use our signup link so that Nerd Vittles gets a referral credit for your signup. Once your account is set up, you'll need to set up a SIP SubAccount and, for Authentication Type, choose Static IP Authentication and enter your Incredible PBX 2021 server's public IP address. For Transport, choose UDP. For Device Type, choose Asterisk, IP PBX, Gateway or VoIP Switch. Order a DID in their web panel, and then point the DID to the SubAccount you just created. Be sure to specify **atlanta1.voip.ms** as the POP from which to receive incoming calls. In the Incredible PBX GUI, be sure to enable the VoIP.ms trunk.

Configuring V1VoIP for Incredible PBX 2021

To sign up for V1VoIP service, sign up on their web site. Then login to your account and order a DID under the DIDs tab. Once the DID has been assigned, choose View DIDs and click on the Forwarding button beside your DID. For Option #1, choose Forward to IP Address/PBX. For the Forwarding Address, enter the public IP address of your server. For the T/O (timeout) value, set it to 20 seconds. Then click the Update button. Under the Termination tab, create a new Endpoint with the public IP address of your server so that you can place outbound calls through V1VoIP. In the Incredible PBX GUI, be sure to enable all of the V1VoIP trunks.

Configuring Anveo Direct for Incredible PBX 2021

To sign up for Anveo Direct service, sign up on their web site and then login. After adding funds to your account, purchase a DID under Inbound Service -> Order DID. Next, choose Configure Destination SIP Trunk. Give the Trunk a name. For the Primary SIP URI, enter

$[E164]$@server-IP-address. For Call Options, select your new DID from the list. You also must whitelist your public IP address under Outbound Service -> Configure. Create a new Call Termination Trunk and name it to match your server. For Dialing Prefix, choose six alphanumeric characters beginning with a zero. In Authorized IP Addresses, enter the public IP address of your server. Set an appropriate rate cap. We like $0.01 per minute to be safe. Set a concurrent calls limit. We like 2. For the Call Routing Method, choose Least Cost unless you're feeling extravagant. For Routes/Carriers, choose Standard Routes. Write down your Dialing Prefix and then click the Save button.

Before you can make outbound calls through Anveo Direct from your PBX, you first must configure the Dialing Prefix that you wrote down in the previous step. Log into the GUI as admin using a web browser and edit the *Anveo-Out* trunk in *Connectivity -> Trunks*. Enable the Trunk. Then click on the *custom-Settings* tab and replace **anveo-pin** with your actual Dialing Prefix. Click *Submit* and *Apply Config* to complete the setup. In the Incredible PBX GUI, be sure to enable all of the remaining Anveo trunks.

By default, incoming Anveo Direct calls will be processed by the Default inbound route on your PBX. If you wish to redirect incoming Anveo Direct calls using DID-specific inbound routes, then you've got a bit more work to do. In addition to creating the inbound route using the 11-digit Anveo Direct DID, enter the following commands after logging into your server as root using SSH/Putty:

```
cd /etc/asterisk
echo "[from-anveo]" >> extensions_custom.conf
echo "exten => _.,1,Ringing" >> extensions_custom.conf
echo "exten => _.,n,Goto(from-trunk,\\${SIP_HEADER(X-anveo-e164)},1)" >>
extensions_custom.conf
asterisk -rx "dialplan reload"
```

Configuring Extensions with Incredible PBX GUI

Extensions are created using the Incredible PBX GUI: **Applications:Extensions**. Many SIP phones expect extensions to communicate on UDP port 5060. If this is the case with your SIP phone or softphone, then always create Chan_SIP extensions which communicate on UDP 5060. If your SIP phone or softphone provide port flexibility, then you have a choice in the type of SIP extension to create: Chan_SIP or the more versatile PJSIP (UDP 5061). Just remember to always configure SIP extensions with *NAT Mode=YES* in the *Advanced* tab. If your VoIP phones or softphones support IAX connectivity, you may wish to consider IAX extensions which avoid NAT problems.

When you create a new Extension, a new entry is automatically created in the PBX Internal Directory. If you wish to allow individual users to manage their extensions or use the WebRTC softphone, then you will also have to create a (very) secure password for User Control Panel (UCP) access. Choose **Admin:User Management** and click on the *key icon* of the desired extension to assign a password for UCP and WebRTC access.

Configuring a Desktop Softphone for Incredible PBX

We're in the home stretch now. You can connect virtually any kind of telephone to your new PBX. Plain Old Phones require an analog telephone adapter (ATA) which can be a separate board in your computer from a company such as Digium. Or it can be a standalone SIP device such as ObiHai's OBi100 or OBi110 (if you have a phone line from Ma Bell to hook up as well). SIP phones can be connected directly so long as they have an IP address. These could be hardware devices or software devices such as the YateClient softphone. We'll start with a free one today so you can begin making calls. You can find dozens of recommendations for hardware-based SIP phones both on Nerd Vittles and the PIAF Forum when you're ready to get serious about VoIP telephony.

We recommend YateClient which is free. <u>Download it from here</u>. Run YateClient once you've installed it and enter the credentials for the 701 extension on Incredible PBX. You can find them by running `/root/show-passwords` . You'll need the IP address of your server plus your extension 701 password. In the YateClient, fill in the blanks using the IP address of your **Server**, 701 for your **Username**, and whatever **Password** was assigned to the extension when you installed Incredible PBX. Click OK to save your entries.
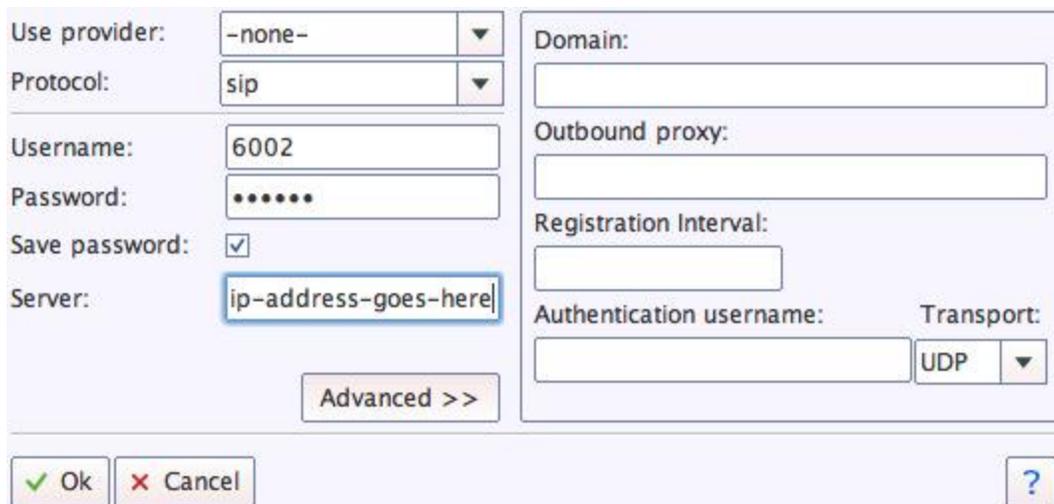
Once you are registered to extension 701, close the Account window. Then click on YATE's Telephony Tab and place some test calls to the numerous apps that are preconfigured on Incredible PBX. Dial a few of these to get started:

```
DEMO - Apps Demo
123 - Reminders
947 - Weather by ZIP Code
951 - Yahoo News
TODAY - Today in History
LENNY - The Telemarketer's Worst Nightmare
```

If you are a Mac user, another great no-frills softphone is <u>Telephone</u>. Just download and install it from the <u>Mac App Store</u>. For Android users, check out the terrific new <u>VitalPBX Communicator</u>. Works flawlessly with Incredible PBX.

Configuring a Softphone Extension on a Smartphone

Adding an Incredible PBX extension to your smartphone gets a little trickier. Whether you're an iPhone or Android lover, all smartphones use batteries, and you don't want to drain your battery by running a softphone as a foreground app all the time. Fortunately, you now have some choices in softphones engineered to work without draining your battery. While they all cost money, it's not much money. We've written about all the choices, and you'll find the links in our Softphone Provider Recommendations on the new Incredible PBX Wiki.

With PJsip extensions, you're not limited to a single phone connection at a time, and we've preconfigured extension 701 to support five simultaneous connections. The setup on the softphone side is simple. For the server, enter the actual IP address of your PBX in the following format: **22.33.44.55:5061**. Then enter 701 for the username and enter the password assigned to the 701 extension on your PBX. When an incoming call arrives, all the phones registered to extension 701 will ring simultaneously. Simply answer the call on the phone that is most convenient.

Configuring Outbound Routes in Incredible PBX GUI

Outbound Routes serve a couple of purposes. First, they assure that calls placed by users of your PBX are routed out through an appropriate trunk to reach their destination in the least costly manner. Second, they serve as a security mechanism by either blocking or restricting certain calls by requiring a PIN to complete the calls. Never authorize recurring charges on credit cards registered with your VoIP providers and, if possible, place pricing limits on calls with your providers. If a bad guy were to break into your PBX, you don't want to give the intruder a blank check to make unauthorized calls. And you certainly don't want to join the $100,000 Phone Bill Club.

To create outbound routes in the Incredible PBX GUI, navigate to **Connectivity:Outbound Routes** and click *Add Outbound Route*. In the *Route Settings* tab, give the Outbound Route a name and choose one or more trunks to use for the outbound calls. In the *Dial Patterns* tab, specify the dial strings that must be matched to use this Outbound Route. *NXXNXXXXXX* would require only 10-digit numbers with the first and fourth digits being a number between 2 and 9. Note that Outbound Routes are searched from the top entry to the bottom until there is a match. Make certain that you order your routes correctly and then place test calls watching the Asterisk CLI to make sure the calls are routed as you intended.

Configuring Inbound Routes in Incredible PBX GUI

Inbound Routes, as the name implies, are used to direct incoming calls to a specific destination. That destination could be an extension, a ring group, an IVR or AutoAttendant, or even a conference or DISA extension to place outbound calls (hopefully with a very secure password). Inbound Routes can be identified by DID, CallerID number, or both. To create Inbound Routes, choose **Connectivity:Inbound Routes** and then click *Add Inbound*

Route. Provide at least a Description for the route, a DID to be matched, and the Destination for the incoming calls that match. If you only want certain callers to be able to reach certain extensions, add a CallerID number to your matching criteria. You can add Call Recording and CallerID CNAM Lookups under the *Other* tab.

Audio Issues with Incredible PBX 2021

If you experience one-way or no audio on some calls, add your external IP address and LAN subnet in the GUI by navigating to *Settings -> Asterisk SIP Settings*. In the *NAT Settings* section, click *Detect Network Settings*. Click *Submit* and *Apply Settings* to save your changes.

Security Considerations with Incredible PBX 2021

Incredible PBX 2021 includes a rock-solid firewall that limits access to preferred providers and individuals whose IP addresses you have whitelisted. Unfortunately, this may not insulate your server from FreePBX 15 irregularities if, in fact, Sangoma's signing key was compromised in the October 2020 Ransomware Attack. Sangoma either doesn't know or isn't telling. Keep in mind that Sangoma didn't mention the October breach either until someone else exposed it. Sangoma's latest press release is available here.

The good news is Incredible PBX 2020 and 2021 platforms include a unique ClearlyIP feature that lets us manage which modules and versions can be installed. It works exactly like what ClearlyIP has documented in their must-read blog post, and we've built a locked version that rolls back all of the modules to dates before the Sangoma breach. The good news is, with Incredible PBX, you don't have to jump through all the hoops covered in the ClearlyIP article to fully insulate your server from the Ransomware breach. We've done the work for you.

For those with mission-critical platforms, we'd recommend immediate implementation of what follows. For everyone else, it's your choice whether to wait and see if there is a breach of the signing certificate with malicious modules. If you opt to wait and see, MAKE FREQUENT BACKUPS.

Here's how to roll back all of your modules to dates before the breach. Login to the FreePBX GUI as admin and navigate to *Settings > Advanced Settings*. Drop down to the **Lock Version** field and change **15.19.11.001** to **15.19.11.003**. Save your changes and reload your dialplan. Then use *Module Admin* to roll back any installed modules that are newer than the safe versions shown.

Our extra special thanks goes to Tony Lewis and the ClearlyIP team for providing this invaluable resource. Somehow we knew it would come in handy sooner or later. Unfortunately, that day has come.

Adding Incredible PBX 2021 to an OpenVPN Network

We previously have documented the procedure for creating an OpenVPN server as well as OpenVPN client templates (.ovpn). If you need a refresher, the <u>tutorial is here</u>. To add your Incredible PBX 2021 server to an existing OpenVPN network, begin by creating an incrediblepbx2021.ovpn template on your OpenVPN server. Be sure to comment out or delete the *setenv* line in the template. Then copy this template to /etc on your Incredible PBX 2021 server. Next, issue the following commands to put the remaining pieces in place:

```
cp -p /root/openvpn-start /etc/.
echo "[Unit]
Description=openvpn2021
ConditionPathExists=/etc/openvpn-start
After=rclocal.service
[Service]
Type=forking
ExecStart=/etc/openvpn-start /etc/incrediblepbx2021.ovpn
TimeoutSec=0
StandardOutput=tty
RemainAfterExit=yes
PermissionsStartOnly=true
SysVStartPriority=99
[Install]
WantedBy=multi-user.target" > /etc/systemd/system/openvpn2021.service
```

Finally, enable the new openvpn2021.service and reboot your server. The OpenVPN IP address should now appear on the LAN line in **pbxstatus**:

```
systemctl enable openvpn2021.service
reboot
```

Incredible PBX 2021 Administration

We've eased the pain of administering your new PBX with a collection of scripts which you will find in the /root folder after logging in with SSH or Putty. Here's a quick summary of what each of the scripts does.

add-fqdn is used to whitelist a fully-qualified domain name in the firewall. Because Incredible PBX 2021 blocks all traffic from IP addresses that are not whitelisted, this is what you use to authorize an external user for your PBX. The advantage of an FQDN is that you can use a dynamic DNS service to automatically update the IP address associated with an FQDN so that you never lose connectivity.

add-ip is used to whitelist a public IP address in the firewall. See the add-fqdn explanation as to why this matters.

del-acct is used to remove an IP address or FQDN from the firewall's whitelist.

admin-pw-change is used to set the admin password for access to the FreePBX/Incredible PBX web GUI using a browser pointed to the local IP address of your server.

apache-pw-change is used to set the admin password for access to Apache/Incredible PBX apps including AsteriDex and Reminders. This provides a password layer of protection for access to these applications.

avantfax-email-change is used to change the destination email address for incoming faxes.

avantfax-pw-change is used to change your admin password for the AvantFax web console.

iaxmodem-restart is used to restart the modems used to send and deliver faxes. The pbxstatus display will tell you whether the IAXmodems are down.

incrediblebackup2021 makes a backup of critical components on your PBX to a tarball saved in /backup. This should be copied to safe location off-site for a rainy day.

incrediblerestore2021 restores a backup file which has been copied to the /backup folder.

ipchecker is a script which deciphers the public IP addresses associated with whitelisted FQDNs created with **add-fqdn** on your server. If any of the addresses have changed, the firewall is restarted after updating the IP addresses. By default, it is executed every 10 minutes by /etc/crontab.

licenses.sh displays the license associated with each of the FreePBX modules on your server.

logos-b-gone removes proprietary artwork from your PBX and is no longer necessary with the included IncrediblePBX FreePBX module.

mime-construct is a command-line utility to send emails with attachments.

neorouter-login is a script to add your PBX to a NeoRouter VPN. Tutorial here.

odbc-gen.sh is a script that was run to generate the ODBC settings for Asterisk. Do NOT use it.

openvpn-start is a script to add your PBX to an existing OpenVPN network using an .ovpn config file. Tutorial here.

pbxstatus displays status of all major components of Incredible PBX 2021.

pptp-install is a script to create a PPTP network connection for your PBX. Tutorial here.

purge-cdr-cel-records removes all CDR and CEL records from the MySQL database.

reset-conference-pins is a script that automatically and randomly resets the user and admin pins for access to the preconfigured conferencing application. Dial C-O-N-F from any registered SIP phone to connect to the conference.

reset-extension-passwords is a script that automatically and randomly resets ALL of the SIP passwords for extensions 701-705. Be careful using this one, or you may disable existing registered phones and cause Fail2Ban to blacklist the IP addresses of those users. *HINT:* You can place a call to the Ring Group associated with all five extensions by dialing 777.

reset-reminders-pin is a script that automatically and randomly resets the pin required to access the Telephone Reminders application by dialing 123. It's important to protect this application because a nefarious user could set up a reminder to call a number anywhere in the world assuming your SIP provider's account was configured to allow such calls.

show-feature-codes is a cheat sheet for all of the feature codes which can be dialed from any registered SIP phone. It documents how powerful a platform Incredible PBX 2021 actually is. A similar listing is available in the GUI at *Admin -> Feature Codes*.

show-passwords is a script that displays most of the passwords associated with Incredible PBX 2021. This includes SIP extension passwords, voicemail pins, conference pins, telephone reminders pin, and your Anveo Direct outbound calling pin (if configured). Note that voicemail pins are configured by the user of a SIP extension the first time the user accesses the voicemail system by dialing *97.

sig-fix disables Module Signature Checking in the FreePBX GUI. This should not be necessary unless you have added or edited FreePBX Modules with missing module signatures.

sms-skyetel is a script to send SMS messages using a Skyetel trunk.

sms-voip.ms is a script to send SMS messages using a VoIP.ms trunk.

sms-blast, **sms-blaster**, and **sms-dictator** are scripts for message blasting. Tutorial here.

switch-to-php5.6 is a script to disable PHP 7.3 and set PHP 5.6 as the default version for your PBX. PHP 5.6 is required to use AvantFax. It is the default configuration for Incredible PBX 2021. The current default PHP version is displayed in the Apache listing of **pbxstatus**.

switch-to-php7.3 is a script to disable PHP 5.6 and set PHP 7.3 as the default version for your PBX. You cannot use AvantFax when PHP 7.3 is the default.

timezone-setup is a script to set the timezone for your PBX.

update-IncrediblePBX is a script that runs the Automatic Update Utility whenever you login to your server as root. These updates typically resolve bugs and security issues with your PBX. Do NOT remove it.

upgrade-asterisk16 is a script that runs on Asterisk 16 platforms to upgrade your PBX to the latest release of Asterisk 16.

upgrade-asterisk18 is a script that runs on both Asterisk 16 and 18 platforms to upgrade your PBX to the latest release of Asterisk 18.

wolfram is a script to deploy Wolfram Alpha on your PBX. Tutorial here.

Forwarding Calls to Your Cellphone. Keep in mind that inbound calls to your DIDs automatically ring all five SIP extensions, 701-705. The easiest way to also ring your cellphone is to set one of these five extensions to forward incoming calls to your cellphone. After logging into your PBX as root, issue the following command to forward calls from extension 705 to your cellphone: `asterisk -rx "database put CF 705 6781234567"`

To remove call forwarding: `asterisk -rx "database del CF 705"`

Keeping FreePBX 15 Modules Current

We strongly recommend that you periodically update all of your FreePBX modules to eliminate bugs and to reduce security vulnerabilities. From the Linux CLI, log into your server as root and issue the following commands:

```
fwconsole ma upgradeall
rm -f /tmp/*
fwconsole reload
/root/sig-fix
/root/sig-fix
```

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BOGO Bonaza: Enjoy state-of-the-art VoIP service with a $10 credit and half-price SIP service on up to $500 of Skyetel trunking with free number porting when you fund your Skyetel account. No limits on number of simultaneous calls plus quadruple data center redundancy assures that you'll never have a failed call. Tutorial and sign up details are here.

The lynchpins of Incredible PBX 2020 and 2021 are ClearlyIP components which bring management of FreePBX modules and SIP phone integration to a level never before available with any other Asterisk distribution. And now you can configure and reconfigure your new Incredible PBX phones from the convenience of the Incredible PBX GUI.

VitalPBX is perhaps the fastest-growing PBX offering based upon Asterisk with an installed presence in more than 100 countries worldwide. VitalPBX has generously provided a customized White Label version of Incredible PBX tailored for use with all Incredible PBX and VitalPBX custom applications. Follow this link for a free test drive!

  Special Thanks to Vitelity. Vitelity is now Voyant Communications and has halted new registrations for the time being. Our special thanks to Vitelity for their unwavering financial support over many years and to the many Nerd Vittles readers who continue to enjoy the benefits of their service offerings. We will keep everyone posted on further developments.
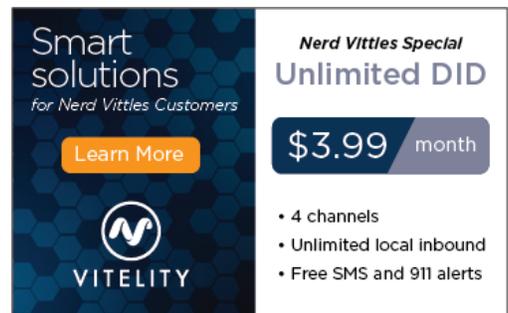
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