VoiceDirector

FAQs

The following FAQs answer most of the common questions about the VoiceDirector service and the compatible devices.

Topics

Setup

Q. What is VoiceDirector and how can my enterprise benefit from it?

A. VoiceDirector is a call management server that provides IP PBX (Public Automatic Branch Exchange) capabilities over a broadband Internet connection. Using VoIP (Voice over IP) functionality, VoiceDirector sends and receives telephone calls over a data network using IP, the Internet Protocol. It offers many of the same calling features as a traditional PBX, including voicemail, Call Waiting, Caller ID, and 3-Way Calling.

VoiceDirector can be implemented as a stand-alone IP-PBX or added to an existing PBX system and provides employees with a familiar calling experience. VoiceDirector is scalable, easy to implement, promotes interoperability, and allows businesses to deploy their communications and IP data solutions on a single, cohesive platform. Best of all, VoiceDirector is designed to drastically reduce the communication costs of small and medium-sized businesses without sacrificing quality of service.

Q. What equipment do I need to get VoiceDirector up and running?

A. You will need:

- A VoiceDirector-approved PC (to act as the server), such as the IBM TM50 or the Dell SC420.
- Internet access (if you will be accessing the VoiceDirector Administrator Interface outside the network.)
- Phones, which include gateways, MTAs IP phones, Wi-Fi phones, and soft phones.
- A VoiceDirector account.
- Phone numbers, or DIDs, (for incoming calls)

Q. Can VoiceDirector be set up behind a NAT?

A. No. VoiceDirector must be set up on a public IP address.

Q. How do I set up VoiceDirector behind my enterprise's firewall?

A. If the VoiceDirector server is installed behind a firewall, you must confirm that the firewall allows the following rules for appropriately routing traffic to and from the VoiceDirector server.

The following ports must be open on your firewall (inbound) in order for VoiceDirector to operate:

PORT	TCP/UDP	SERVICE	DIRECTION
80	TCP	Provisioning	Outbound
80	TCP	Remote Configuration	Inbound
5060	UDP	SIP (signaling)	Inbound/Outbound
20000:22000	UDP	RTP (media)	Inbound/Outbound

TIP: To be able to administer the VoiceDirector network remotely (from outside the network in which the VoiceDirector server resides), port 80 TCP must be open.

VoiceDirector Configuration

License Keys

Q. What types of license keys are available?

A. The table below describes each key type and the corresponding number of allowable ports and concurrent calls.

LICENSE KEY LEVEL	NUMBER OF SEATS (PORTS) ALLOWED	NUMBER OF CONCURRENT CALLS ALLOWED
Bronze	100	10
Silver	150	15
Gold	250	25

Acquiring Accounts and DIDs

Q. Can I reserve a consecutive series of DIDs (e.g., xxx-nnn-4000 through xxx-nnn-4999)?

A. No. We do not currently offer consecutive DIDs.

Q. Can I get multiple accounts for a single VoiceDirector server?

A. VoiceDirector currently handles one account only.

Device Configuration and Troubleshooting

Compatibility

Q. What devices are compatible with VoiceDirector?

A. VoiceDirector supports a variety of SIP-compliant gateways, IP phones, Wi-Fi phones, and PCI cards. Following is an updated listig of VoiceDirector certified devices.

- **Cisco ATA186:** Supported firmware: 3.10, 2.16, 2.15
- **Sipura SPA2000:** Supported firmware: 2.0.13(g)
- **Sipura SPA2100 (router):** Supported firmware: 2.0.1(a)
- **AudioCodes MP-104/FXS/AC/SIP:** Supported firmware: 4.40.193.350
- **AudioCodes MP-104/FXO/AC/SIP:** Supported firmware: 4.40.193.350
- **Koncept KE1021A IP Phone:** Supported firmware: 4.04.00
- **LG LIP6830/LIP6912:** Supported firmware: N/A
- **BCM WLAN-600:** Supported Firmware: WB003A
- **Digium TDM01B TDM400P Card FXO and FXO Module TDM400P**

Device-Specific Information

LG LIP-6812/ LIP-6830

GENERAL

Q. The MSG light blinks, but when I push it, it does not dial voicemail.

A. Make sure to put your voicemail ID (probably your extension) in the **VMS Address** field in the LG LIP phone Web interface.

Q. I configured Call Forwarding on my LG LIP phone, but it doesn't work.

A. Because of security considerations, users on the VoiceDirector network can only forward calls to other extensions within the network and not to other PSTN (Public Switched Telephone Network) numbers (outside the network).

Q. I am unable to access the Web Manager after entering the correct URL string in my browser (http://xxx.xxx.xxx.xxx:8000).

A. Check the **Display Name** parameter configuration on the Web Manager to confirm that it does not have a space. Access to a device's Web Manager may fail if the Display Name parameter is configured with a value that includes a space.

For example: Test Test

Q. Can calls be placed from a LIP Phone to a different hardware product? If so, how?

A. The LIP Phone can communicate to other LIP Phones along with all other VoiceDirector-supported hardware and software clients registered on the same VoiceDirector network, as well as PSTN phones. To place an in-network call, simply dial the VoiceDirector extension of the person you wish to call. To place an out-of-network call, first dial the prefix **9**, followed by the full destination phone number.

Q. Will my LIP Phone work with ISDN, DSL or cable?

A. Yes, they will operate with ISDN, DSL or cable. Please note that the amount of available bandwidth on your network may determine what codec setting must be used on your LIP Phone.

Q. I am using a DSL or Cable-based connection to the Internet. Do I need additional accessories to start making calls?

A. If you are using a PPPoE connection to access the Internet (such as cable modem or DSL), you will need to connect the LIP Phone to a router, switch, or hub, which then connects to the cable or DSL modem.

Q. When I call the United States, the call quality is very good, but I experience a delay when calling other countries. Why?

A. This is most likely occurring in the local phone system and is independent of the product.

Some call delay issues can be caused by limitations of voice traffic travel to the terminating carrier at the destination's end. VoiceDirector cannot control this.

Q. My audio cuts in and out or is distorted. What can I do to help fix this problem?

A. Check the codec settings on your LIP Phone's VoIP Config page (in the Web Manager); they may need to be adjusted in order to achieve better audio quality. If you have low bandwidth, set your **Codec Priority 1** (primary) setting to **G.729** codec. The **Codec Priority 2** (secondary) setting can then be set to either **G.711** codec or none.

If this does not help, you may need to readjust the **Jitter Buffer Bounds** on the QoS Configuration page of the LIP Web Manager. Please note that the following default values apply:

Minimum Delay: 30 ms

Normal Delay: 60 ms

MAXimum Delay: 120 ms

Please note that while setting a higher overall jitter buffer range may decrease the risk of audio packet loss, it may increase the chance for audio delays.

If you are still encountering problems with audio, also note that call quality may be affected by the available bandwidth of the network to which your device is connected.

FUNCTIONALITY

Q. What bandwidth is required to use the LIP Phone?

A. Your bandwidth requirement depends on the type of codec your LIP Phone is set up to use. For calls set up with the G.711 codec, you will need a minimum of 160Kbps of bandwidth on a single call. If you are using the G.729 codec, you will need a minimum of 53Kbps on a single call.

Q. I cannot log into my voicemailbox after calling it using the MSG button on the LIP-6830 phone, or any programmed Speed Dial button. Why?

A. If you use Speed Dial or the **MSG** button to dial a phone number that is answered by an automated menu (i.e., an Interactive Voice Response, or IVR), the call may not connect successfully. It is recommended that Speed Dial keys do not get programmed with these types of phone numbers.

Q. I have trouble logging into my voicemailbox after I call it using the MSG button on the LIP-6830 phone.

A. Verify that you have programmed the **MSG** button to properly dial out to your voicemail. You can program the button by entering your voicemail access number in the **VMS Address** field in the VoIP Configuration page of the Web Manager.

Q. I have more than one line configured on my LIP-6830 phone. How can I set the MSG button on my LIP-6830 phone to access voicemail for my other lines?

A. The **MSG** button on the LIP-6830 phone will only dial out through Line 1. Voicemail cannot be accessed for accounts other than the one that is configured on Line 1 when using this button.

SETUP/CONNECTIVITY

Q. I deleted all of the entries in my Routing Table and now I can't place any calls.

A. Some entries within the Routing Table contain settings that are necessary to place outbound calls. If these values have been deleted, you must reload your default configuration in order to restore those entries.

Note: Restoring defaults will remove all other configuration from your device, including proxy address and account information. It is strongly recommended that all configuration parameters be noted prior to loading the default configuration.

Q. I set DHCP to ON, but my LIP-6812/LIP-6830 cannot obtain correct network settings.

A. Check the following:

- Make sure your network supports DHCP.
- Make sure your RJ-45 LAN cable is securely connected to both the LIP-6812/LIP-6830 and to the LAN, hub, or router.
- Also, make sure that all network ports, routers, or hubs are on.

Q. My analog phone's keypad does not have a period. How do I enter the periods for my IP address and netmask?

A. Use the star key (*) to enter the periods.

Q. When I enter a static IP address into the LIP Phone and restart the unit, the static IP address that I entered is not saved. What should I check?

A. Make sure that the **Network Selection** radio button on the LAN Configuration page of the Web Manager is set to **DHCP**.

Q. I changed the login password on my LIP-6812/LIP-6830, and I forgot the new password.

A. To recover the default password on your device, you must load the default configuration.

[***Note***: Restoring defaults will remove all other configuration from your device, including proxy address and account information.]

If you are also locked out from accessing the LIP Phone's LCD menu, then access the Web Manager page, and log in as **common**. Note that the default password for both is **n2p**.

Q. When I attempt to access the Web Manager, I am prompted for a user name and password. However, when I enter them and press the OK button, the login window reappears.

A. Make sure both the user name and password are correct (the user name and password are case sensitive).

Q. If I already have a LIP Phone and want to add more, what are the installation issues involved?

A. You may add additional LIP Phones to a network just as you would add additional computers to a network. Make sure there is enough bandwidth present and that the appropriate IP addresses are available.

TROUBLESHOOTING

Q. When I attempt to place a call from my LIP Phone to another phone number, I get a busy signal. What could be wrong?

A. There could be several possible causes for this problem:

- Verify that the **Proxy Address** configured for the particular Line (1 or 2) in the VoIP Configuration page of the LIP Phone's Web Manager is correct.
- Verify that the **SIP Username** parameter in the VoiceDirector Web Manager matches the **Name** and **Authentication Name** parameters in the LIP Phone's Web Manager.
- Verify that the **SIP Password** parameter in the VoiceDirector Web Manager matches with the **Authentication Password** parameter on the LIP Phone's Web Manager.
- Verify that the Device and User profiles for your LIP Phone are properly associated together on the VoiceDirector server's Web Manager.
- If your device is set up behind a router, make sure that the **Behind NAT** radio button is set to **Yes** in the Device profile within VoiceDirector.
- If placing out-of-network calls, make sure that your User profile on VoiceDirector has an Account Number associated with it.
- Verify that the account you are calling from has sufficient funds (if placing out-of-network calls).
- Make sure the codec settings on your device are properly set to support the codec your VoiceDirector server uses to set up calls.

  [For example, if the VoiceDirector server to which the LIP Phone is registered uses G.711 only, then the device must have G.711 set as Codec Priority 1.]

Q. Even though I am able to place calls on my LIP Phone, I am unable to receive incoming calls. What could be wrong?

A. There could be several possible causes for this problem:

- Make sure that the device shows an Online (Green) status in the Real Time Display page of the VoiceDirector Web Manager.

- Make sure your VoiceDirector SIP Username and SIP Password are not set on another device.

If the other party is dialing a Hunt or Fork Group extension number to reach you, verify first that you are on the group list.

If you are still unable to receive calls to your phone, reboot your device so that it re-registers with VoiceDirector.

Q. When I call the United States, the call quality is very good, but I experience a delay when calling other countries. Why?

A. This is most likely occurring in the local phone system (of the destination phone number) and cannot be controlled through VoiceDirector.

Q. When I make a call from one LIP Phone to another, the other party cannot hear my voice. What could be the problem?

A. There could be various reasons for this problem:

- First, make sure that both units have been upgraded to the latest firmware.
- One of the phones could be behind a firewall that is blocking the RTP Port used on the call. Check the RTP port settings on the phone behind firewall and make sure that port is not blocked. The RTP port used by the phone can also be changed if required.

  To change the RTP port, follow the steps below:

  1. Log into the Web Manager and access the VoIP Configuration page.
  2. In the **RTP Start Port** field, enter the port number you wish to use for calls.

     NOTE: If your LIP Phone is set behind a firewall, verify that the port number you entered is open on that firewall.

Q. My audio is distorted or choppy. What can I do to help fix this problem?

A. You can adjust the codec settings on your LIP Phone to achieve better audio quality. If your network has limited bandwidth, access the VoIP Config page of the Web Manager, and set your **Codec Priority 1** (primary) setting to **G.729**. The **Codec Priority 2** (secondary) setting can then be set to either **G.711** codec or none.

If you are still encountering problems with audio, please note that call quality may also be affected by the available bandwidth of the network to which your device is connected.

Q. Call Transferring is not working when performing the steps in the User Guide.

A. If your LIP Phone is connected to a VoiceDirector server, Call Transfer must be performed as such:

1. While the call is in progress, press the pound (#) key.
2. After the prompt is played, enter the extension to which you would like to transfer the call.

Q. The Input IP Address and Input URL call modes do not work when they are used.

A. These call modes are not supported if your LIP Phone is registered on a VoiceDirector network. The only call mode that is supported from the [Cmod] (Call Mode) menu is the default calling option, "Input dial number".

Q. My LIP Phone has a tendency to freeze during regular use. Why?

A. Check your Network Time Configuration settings. The LIP Phone may lockup momentarily if it is configured with certain SNTP Server Address values. For best results, it is suggested that you configure your LIP Phone with any of the following time server addresses:

- ntp.nasa.gov
- clock.via.net
- tick.ucla.edu

Q. When I try to dial out automatically from an entry listed in my Call Log, I get a busy signal.

A. If the phone number in the Call Log is not a valid extension within the VoiceDirector network, then the call will fail. In order to place an out-of-network call, you must press the **9** key prior to entering the destination phone number.

Q. When do I know that my device has successfully registered with the VoiceDirector server and is ready to make phone calls?

A. Go to the VoIP Config page of the LIP Phone's Web Manager, and enter a valid **Name**, **Authentication Name**, **Authentication Password**, and **Proxy Address**.

The device should reboot and attempt to log into the VoiceDirector server. If the **Registration Status** in the Web Manager reads **OK**, then the device has successfully registered with the VoiceDirector server and is ready to make calls.

MAX 410/420/430 SIP

GENERAL

Q. Can calls be placed from one MAX SIP device to a different hardware product? If so, how?

A. The MAX 410/420/430 SIP devices can communicate with each other, with all other VoiceDirector-supported hardware and software clients, as well as with PSTN phones. To place an in-network call, simply dial the VoiceDirector extension of the person you wish to call. To place an out-of-network call, first dial the prefix **9**, followed by the full destination phone number.

Q. Will my MAX 410/420/430 SIP device work with ISDN, DSL or cable?

A. Yes, all MAX 4x0 SIP devices operate with ISDN, DSL, or cable. The amount of available bandwidth will determine how many simultaneous calls can be made from the MAX device.

Q. I am using a DSL or cable-based connection to the Internet. Do I need additional accessories to start making calls?

A. If you are using a PPPoE connection to access the Internet (such as a cable modem or DSL), you will need to connect the MAX device to a router, switch, or hub, which then connects to the cable or DSL modem.

Q. When I call the United States, the call quality is very good, but I experience a delay when calling other countries. Why?

A. This is most likely occurring in the local phone system and is independent of the product.

Sometimes call delay issues are caused by limitations of voice traffic travel to the terminating carrier at the destination's end. This is not something that can be controlled.

Q. My audio cuts in and out or is distorted. What can I do to help fix this problem?

A. The codec settings on your MAX 410/420/430 SIP may need to be changed in order to achieve better audio quality. If you have low bandwidth, set your **Codec Priority 1** (primary) setting to the **G.729 codec** within the VoIP Config page of the MAX Web Manager. The **Codec Priority 2** (secondary) setting can then be set to either **G.711 codec** or none.
If this does not help, you may need to readjust the Jitter Buffer Bounds on the Network and Voice Config page of the MAX Web Manager. Please note that the following values apply:

- **jitter buffer value 1** --> jitter delay 60 ms
- **jitter buffer value 2** --> jitter delay 90 ms
- **jitter buffer value 3** --> jitter delay 120 ms (Default for Voice Lower Bound)
- **jitter buffer value 4** --> jitter delay 150 ms
- **jitter buffer value 5** --> jitter delay 180 ms (Default for Voice Upper Bound)
- **jitter buffer value 6** --> jitter delay 210 ms
- **jitter buffer value 7** --> jitter delay 240 ms
- **jitter buffer value 8** --> jitter delay 270 ms
- **jitter buffer value 9** --> jitter delay 300 ms

Please note that while setting a higher overall jitter buffer range may decrease the risk of audio packet loss, it may also increase the chance for audio delays.

If you are still encountering problems with audio, be aware that call quality may be affected by the available bandwidth of the network in which your MAX device is connected.

Q. When I attempt to place a call from my MAX 410/420/430 SIP to another phone number, I get a busy signal. What could be wrong?

A. There could be several possible causes for this problem:

- Make sure the **SIP User Name** and **SIP Password** configured on VoiceDirector matches with the **Name**, **Authentication Name**, and **Authentication Password** configured on the MAX device's Web Manager.
- Make sure that a User profile exists on VoiceDirector and is associated with the Device profile.
- If your device is set up behind a router, make sure that the **Behind NAT** radio button is set to **Yes** in the Device profile within VoiceDirector.

If you are trying to place an out-of-network call, verify that your User profile on VoiceDirector is associated with an Account Number.

- Verify that the codec settings on your device are properly set to support the codec your VoiceDirector server uses to set up calls.

  For example: If the **System-wide Codec** parameter on the Global page of the VoiceDirector Web Manager is set to **G.711**, then the MAX device registering to it must have **G.711** set as **Codec Priority 1**.

Q. Even though I am able to place calls on my MAX 410/420/430 SIP, I am unable to receive incoming calls. What could be wrong?

A. There could be several possible causes for this problem:

- Make sure that your MAX device shows an **Online** (Green) status in the Real Time Display page of the VoiceDirector Web manager.
- Make sure your VoiceDirector **SIP Username** and **SIP Password** are not set on another device.
- If the other party is dialing a Hunt or Fork Group extension number to reach you, verify first that you are on the group list.

NOTE: If you are running software version v1.1.10 on your MAX, you MUST make sure that all ports are configured with an **Authentication Name** and **Authentication Password**. If you only have one such account, you must configure your device in the following manner:

1) Configure each port that you will use on your device with the valid **Authentication Name** and **Authentication Password**.

2) On the ports that will not be used on your device, enter invalid values for the **Authentication Name** and **Password** fields.]

3) Save the values and reboot the device.

Q. When do I know that my device has successfully registered with the VoiceDirector server and is ready to make phone calls?

A. Enter a valid **Name**, **Authentication Name**, **Authentication Password**, and **Proxy Address** through the VoIP Config page of the MAX device's Web Manager, save these settings and reboot your device.

After rebooting, the device should attempt to log into the VoiceDirector server. If the Registration Status in the MAX Web Manager reads **OK**, then the device has successfully registered with the VoiceDirector server and is ready to make calls.

Q. When I make a call from one MAX 410/420/430 SIP to another MAX 410/420/430 SIP, one side cannot hear the communication. What could be the problem?

A. There could be various reasons for this problem:

First, make sure that both units have been upgraded to the latest firmware. (Check the software version on your device by accessing the System Information page on the Web Manager, or log into the command line interface and examine the **/SYSTEM/VERSION** menu option.) Go to www.voicedirector.net to confirm if there is a new release after your version.

If upgrading the software version does not solve the problem, one of the phones could be behind a firewall that might be blocking RTP traffic. Check the RTP port settings on the phone behind the firewall and make sure that port is not blocked. The RTP port used by the phone can be changed if required.

Q. Will I still be billed for a call if the other party does not answer?

A. No. You should not be billed for a call that is not completed.

FUNCTIONALITY

Q. What does FXS mean?

A. FXS stands for "Foreign Exchange Station". This allows you to plug standard analog phones into a MAX FXS unit. The FXS version of the MAX does not support inbound PSTN calling, but you can receive calls (via ***72**) from other MAX units. It is possible to attach FXS lines to a PBX through an analog trunk card (COIC/COIB). The MAX 4, MAX 8, MAX 8 Plus and MAX 8/16 all feature FXS versions.

Q. What does FXO mean?

A. FXO stands for "Foreign Exchange Operator". This allows you to connect the MAX to a PBX through an analog line card (SLIC/SLIB). With an FXO unit, inbound PSTN calls and remote calling are supported. The MAX 8 Plus and MAX 8/16 both feature FXO versions.

Q. What bandwidth is required to use the MAX 410/420/430 SIP?

A. Your bandwidth requirement will depend on the type of codec your MAX is set up to use.

For calls set up with the **G.711** codec, you will need a minimum of 160Kbps of bandwidth on a single call.

If you are using **G.729** codec, you will need a minimum of 53Kbps per single call.

Q. How can I set a longer pause time for placing a call?

A. The default pause time is three (3) seconds. If you would like to set a longer pause time between dialing a number and placing the call, you will need to telnet into your MAX gateway and configure the FXS, FXO, or IVR port to a longer second value. For more information, refer to the *MAX 410/420/430 Command Reference Guide*.

Q. What will happen if all of the lines from the MAX are in use? Can I still place a call?

A. The PBX can be programmed so that if all of the MAX lines are in use, calls will be routed through the PSTN.

Q. What functionality does the IVR present and how is it programmed?

A. The IVR may be programmed to play a greeting and/or request certain information (i.e. a password) when a port is accessed. The IVR function can be configured through the MAX Web Manager, or through a telnet connection by using the commands found in the *MAX 410/420/430 Command Reference Guide*.

Q. If I already have a MAX and want to add more, what installation issues are involved?

A. You would add additional MAX units to a network just as you would add additional computers to a network. Make sure there is enough bandwidth present and the appropriate IP addresses are available.

SETUP/CONNECTIVITY

Q. I deleted all of the entries in my Routing Table and now I can't place any calls.

A. Some entries within the Routing table contain settings that are necessary to place outbound calls. If these values have been deleted, you must reload your default configuration in order to restore those entries.

Note: Please take note that restoring defaults will remove all other configuration from your device, including proxy address and account information. It is strongly recommended that all configuration parameters are noted prior to loading the default configuration.

Q. During setup, how do I enter the periods in my IP address and netmask through an analog phone?

A. Use the star (*) key on the phone to enter a period.

Q. When I enter a static IP address into the MAX and restart the unit, the static IP address that I entered is not saved. What should I check?

A. Make sure that the **Use DHCP** checkbox is unchecked in the MAX Web Manager. If you entered the IP address through an analog phone, make sure you press the pound (#) key before hanging up.

Q. When I attempt to access the MAX Web Manager, I am prompted for a user name and password. When I enter them and press OK, the password page reappears. What am I doing wrong?

A. Please make sure that both the user name and password are correct.

Note: The user name and password are case sensitive.

Q. When would I need to use the serial port on the back of the MAX?

A. If you have a MAX 410 SIP unit, you will need to utilize the serial port for the initial configuration of the LAN settings since you cannot connect a standard analog phone directly to an FXO port.

Q. When I attempt to connect to my MAX through a serial cable connection, I do not see any text, or the text that appears is distorted.

A. Make sure the connection settings are as follows:

Baud rate = 19200
Parity = None
Character Size = 8
Stop Bit = 1
Flow Control = None

Make sure the serial cable is connected and fastened to both the MAX and your computer.

TROUBLESHOOTING

Q. When a call is placed remotely into the MAX (FXO), the phone line does not always disconnect after the caller hangs up. What can be done to change this?

A. Most likely, you will need to change the **Ring Off Detect** setting on the MAX. Please see the technical bulletin on this subject for more information.

Q. My network supports DHCP, but I am having trouble obtaining the correct network settings for the MAX device. What should I check?

A. Please make sure that DHCP is enabled on the MAX device. Also check that the RJ-45 LAN cable is connected securely to the MAX and the LAN/hub/router. Make sure that all network ports, routers, or hubs are live.

Q. When I dial the extension that I routed to port 1, it rings on port 2, and when I dial the extension I routed to port 2, it rings on port 3. What is wrong?

A. In the Routing Table Configuration page, make sure you routed each extension to the logical port number of the desired port, not the labeled or physical port number.

Q. When I attempt to make a call from either an FXS or FXO port, I immediately get disconnected. What should I do?

A. Use the ping command to determine that the MAX still has a live connection to the LAN.

Features

Inbound and Outbound Calling

Q. I have put the account number and phone numbers in the server, but I'm not getting any calls.

A. Make sure the phone number is entered correctly. For example, if you purchased the number 555-1212 in the 973 area code, check that the number you entered was 19735551212.

Call Groups (Fork Groups and Hunt Groups)

Q. What are call groups?

A. VoiceDirector offers the possibility of associating multiple users and device ports into call groups. Call groups can have an extension associated with them, as well as a voicemail box or a phone number.

Incoming calls to the group extension can ring to multiple users simultaneously or one at a time, depending on the call group type.

There are two (2) types of call groups:

- **Hunt Groups**—A hunt group consists of a series of phone extensions or external phone numbers. When a call comes in, it is routed to the extension/phone number of the first member listed in the hunt group. If that line is busy or if there is no answer after a defined amount of time, the call is routed to the next member of the hunt group, and so on. If the last member of the hunt group does not answer the call, it is forwarded to the voicemail box assigned to the hunt group.

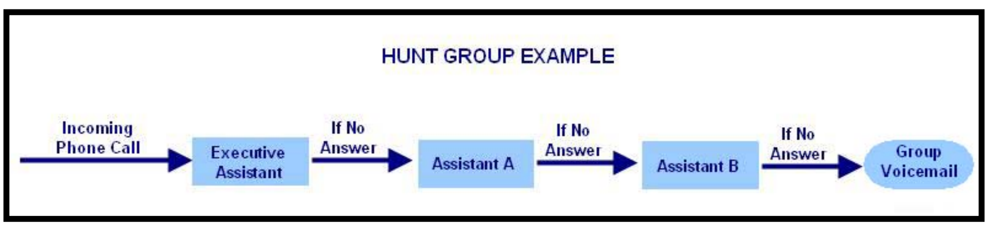
- **Fork Groups**—A fork group consists of at least two phone extensions/external phone numbers. When a call comes in, it is routed to all members the fork group at the same time. When one member answers the call, the other members' phones stop ringing. If none of the members answers the call, it is forwarded to the voicemail box assigned to the fork group.

Q. What are external call groups?

A. External call groups are fork groups or hunt groups in which one of the "member" phone numbers is a number outside the VoiceDirector network. For example, you can set up a fork group that will direct incoming calls to a user's office extension (inside the network) and his cell phone (outside the network) simultaneously.

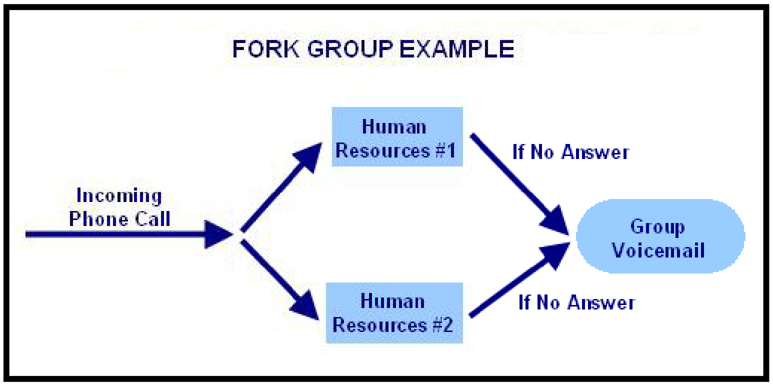
Q. How do I use call groups?

A. *The following is an example of how you could use a **hunt group**:*
Your company's CEO has one executive assistant and two assistants. You can set up a hunt group to route incoming calls to the executive assistant first, then assistant A, and then assistant B. If the executive assistant doesn't answer, the call bounces to assistant A. If she doesn't answer, the call then bounces to assistant B. If no one answers, the call is directed to the hunt group's voicemailbox.



*The following is an example of how you could use a **fork group**:*
Your company's Human Resources department consists of two people, and they would like to receive calls simultaneously. You can set up a fork group to route incoming calls to both people at the same time. When one of the extensions answers, the other stops ringing. If no one answers, the call is directed to the fork group's voicemailbox.

Refer to the image on the following page.



Voicemail

Q. I log in to voicemail, but when I put in my voicemail password, the announcement says the password is wrong.

A. The voicemail password for telephone voicemail access is 1234 and can be changed through the telephone. The password you are given by your VoiceDirector System Administrator is the password for logging in through the Web interface.

Q. I want to quickly check my messages, but the voicemail system takes too long reporting information.

A. Press 1 at any time to bypass the message information and listen to the voicemail directly.

Q. I left a voicemail for someone, and the person did not receive the message.

A. The VoiceDirector System Administrator may choose to ignore voicemails that are shorter than a certain length. This is set in the Global Settings page, in the **Min msg length (sec)** field. If the length of the message is less than the number of seconds indicated in that field, the message is ignored (deleted).

Reports

Real Time Reporting

Q. How do I know if an extension is offline?

A. Log into the VoiceDirector Administrator Interface, and click the **Real Time Display** icon or link. This page displays each extension and its status. A red dot indicates if an extension is offline. Some devices also have an online/offline indicator LED or display.

Q. If a device is offline, can it still place and receive calls?

A. When an extension is offline, it cannot receive calls, but it may be able to place calls, depending on whether the extension is logged in.

Support

Q. How do I contact support?

A. If you are experiencing technical issues with the VoiceDirector software, please refer to the VoiceDirector support page at **http://www.voicedirector.net/** for detailed troubleshooting, software upgrades, and other information**,** or contact your reseller for assistance