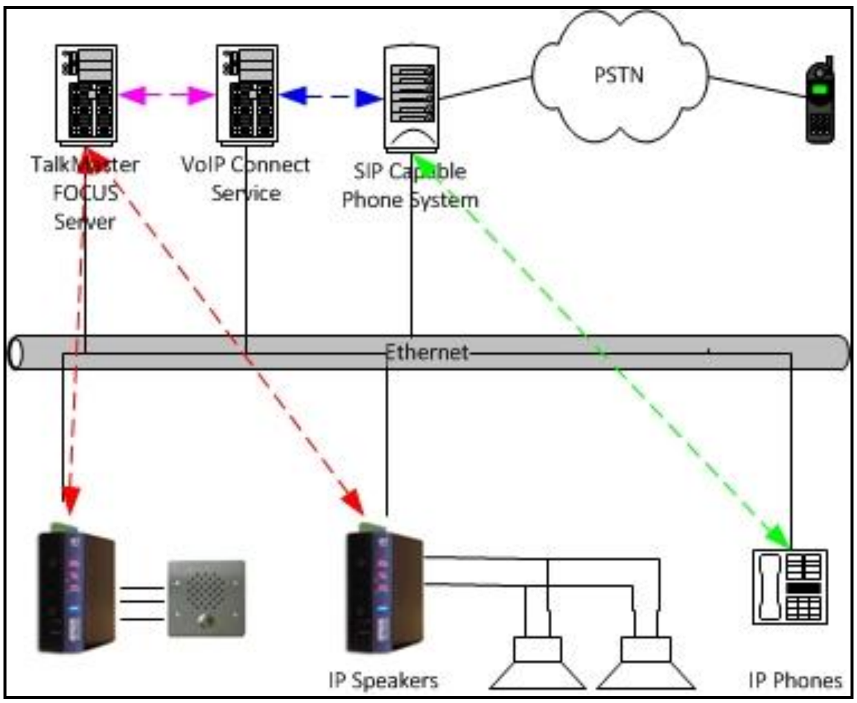
TalkMaster™ FOCUS

VoIP Connect

Reference Manual



DIGITAL ACOUSTICS

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Getting Started

Welcome

Welcome to **TalkMaster FOCUS VoIP Connect**.

VoIP Connect provides the ability to:

- Initiate a 1-way audio page from a phone on a SIP based phone system to one or more IP Endpoints
- Initiate a 2-way call from a phone on a SIP based phone system to a single IP Endpoint
- Initiate a 2-way call from a single IP Endpoint to a phone on a SIP based phone system
- During a 2-way call, the phone on the SIP based phone system can activate IP Endpoint's Relay to open a door

VoIP Connect is comprised of the **VoIP Connect Service** and the **VoIP Connect Configuration Console**. The **VoIP Connect Service** uses the configuration information from the **VoIP Connect Configuration Console** to create a SIP gateway between a SIP based phone system and the IP Endpoints.

Release Notes

V5.1.2 June 2016

- Many stability enhancements for the VoIP Connect Service

V5.1 November 2015

- Initial Release as a Windows Service (replaced the SIP Media Gateway Console)

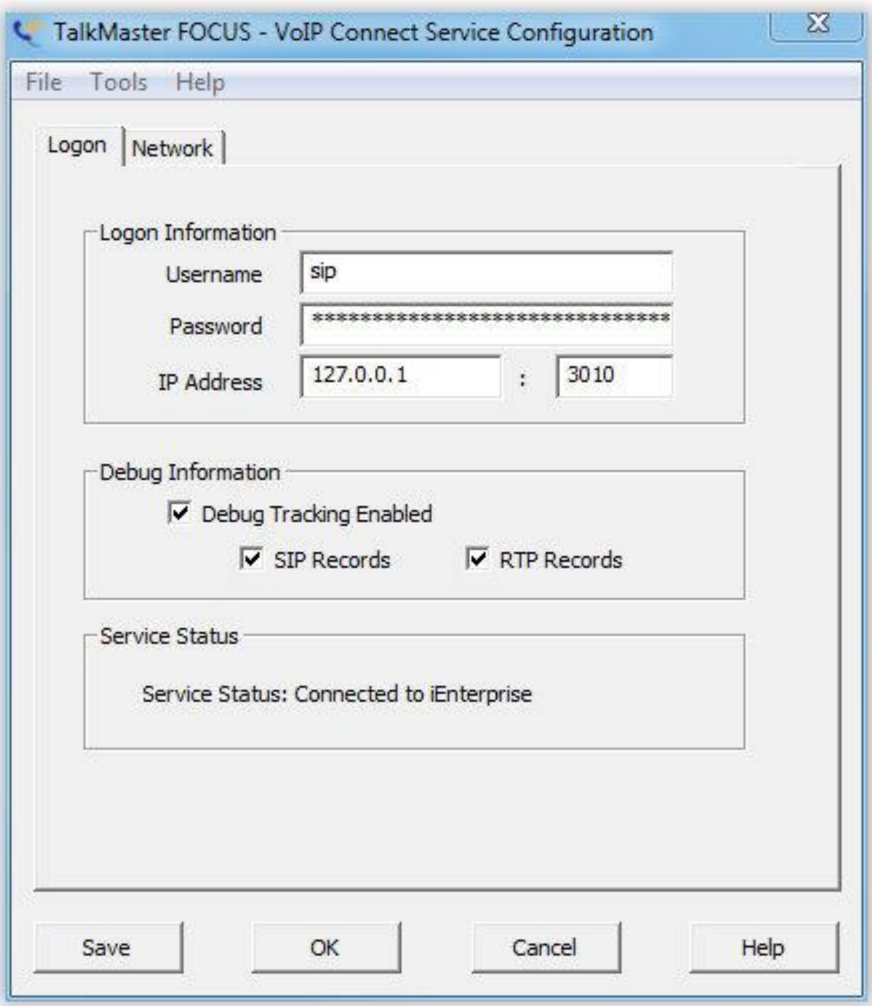
VoIP Connect Service

VoIP Connect Service Logon

The VoIP Connect Service typically runs on the same machine as the **TalkMaster FOCUS Server** service. The **VoIP Connect Service** must logon to the **TalkMaster FOCUS Server** in order to communicate with it, so an Operator UserID and Password needs to be supplied to it as a one-time setup. Please Note that this Operator will always be logged on.

To configure the logon information for the **TalkMaster FOCUS VoIP Connect Service**, launch the **Voip Connect Service Configuration** application from the **Windows Start** menu **All Programs --> TalkMaster FOCUS --> VoIP Connect --> VoIP Connect Service Configuration**.

This console can also be used to display the current status of the service or to restart the service.



Logon tab

- **Username / Password** - Enter the **Username** and **Password** fields that will be used by the **VoIP Connect Service** to login to the **TalkMaster FOCUS Server** service. The **Username** must be defined in the **Admin Console** under the **Operators** tab
- **IP Address** - Enter the IP Address of the **TalkMaster FOCUS Server** and the associated **Console Port** as defined in the **Admin Console**. The **Console Port** defaults to 3010
- **Debug Tracking** - This should be left unchecked unless instructed by Digital Acoustics Technical Support
- **Server Status** - The Service Status area will display the current connection status of the service

Save - Once the above changes have been made, press this button to apply the changes:

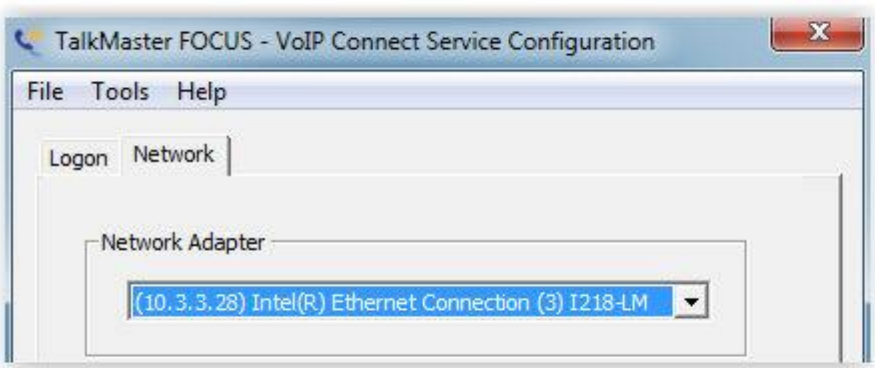
- If the **Server Status** indicates that the **Alert Management Service** is running, but not connected to the **TalkMaster FOCUS Server**, it will try to logon immediately and the **Server Statu**s should change to **Connected to iEnterprise**
- If the **Server Status** indicates that the **Alert Management Service** is not running, then select menu option **Tools --> Restart Service** to connect with the updated information

OK - Press this button to save the changes and exit

Cancel - Press this button to ignore the unsaved changes and exit

VoIP Connect Service Network Adapter

If the machine that the VoIP Connect Service is running on has multiple network adapters, select the Network Adapter that can communicate with the SIP based phone system.

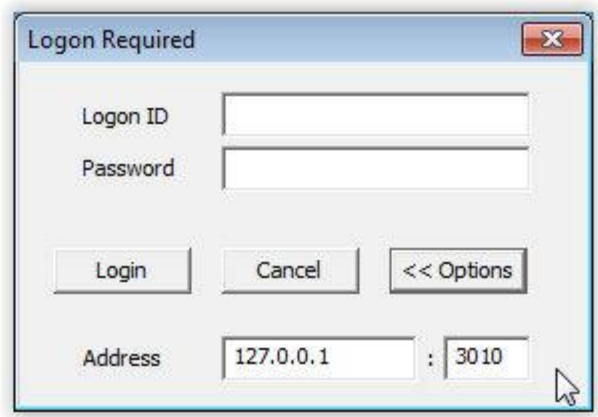


VoIP Connect Configuration Console

Setting Up Extensions

Logon

To start the VoIP Connect Configuration Console, open the **Windows Start** menu --> **All Programs --> TalkMaster FOCUS --> VoIP Connect --> VoIP Connect Service Configuration**. The following logon dialog is displayed:

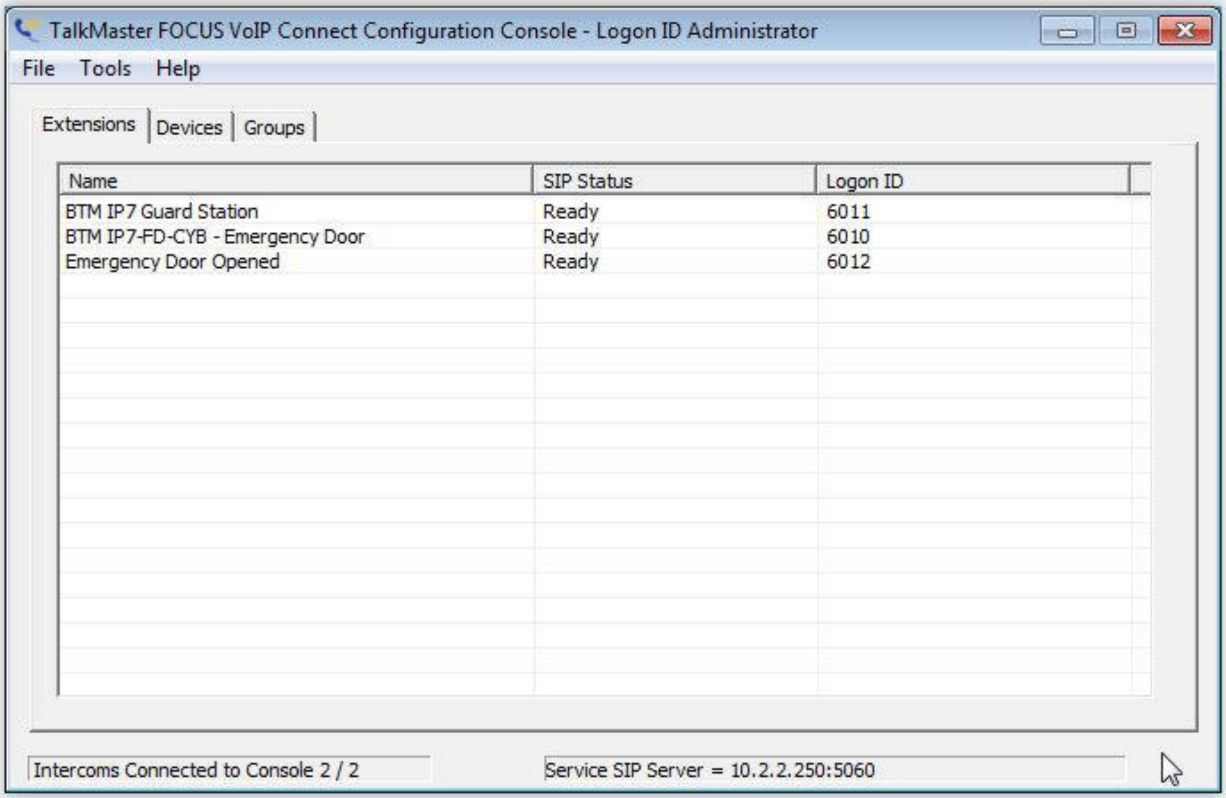


- **Logon ID** - Enter the Logon ID as configured in the **Admin Console --> Operators** tab
- **Password** - Enter the associated password for the Logon ID
- **Address** - Enter the IP Address and Console Port of the TalkMaster FOCUS Server.

In the Admin Console on the Operators tab, check the **Allow Multiple Logons** checkbox for the Operator ID that is used for administering TalkMaster instead of creating a separate UserID for each configuration application

Extensions tab

The main screen allows you to see three different tabs of information.



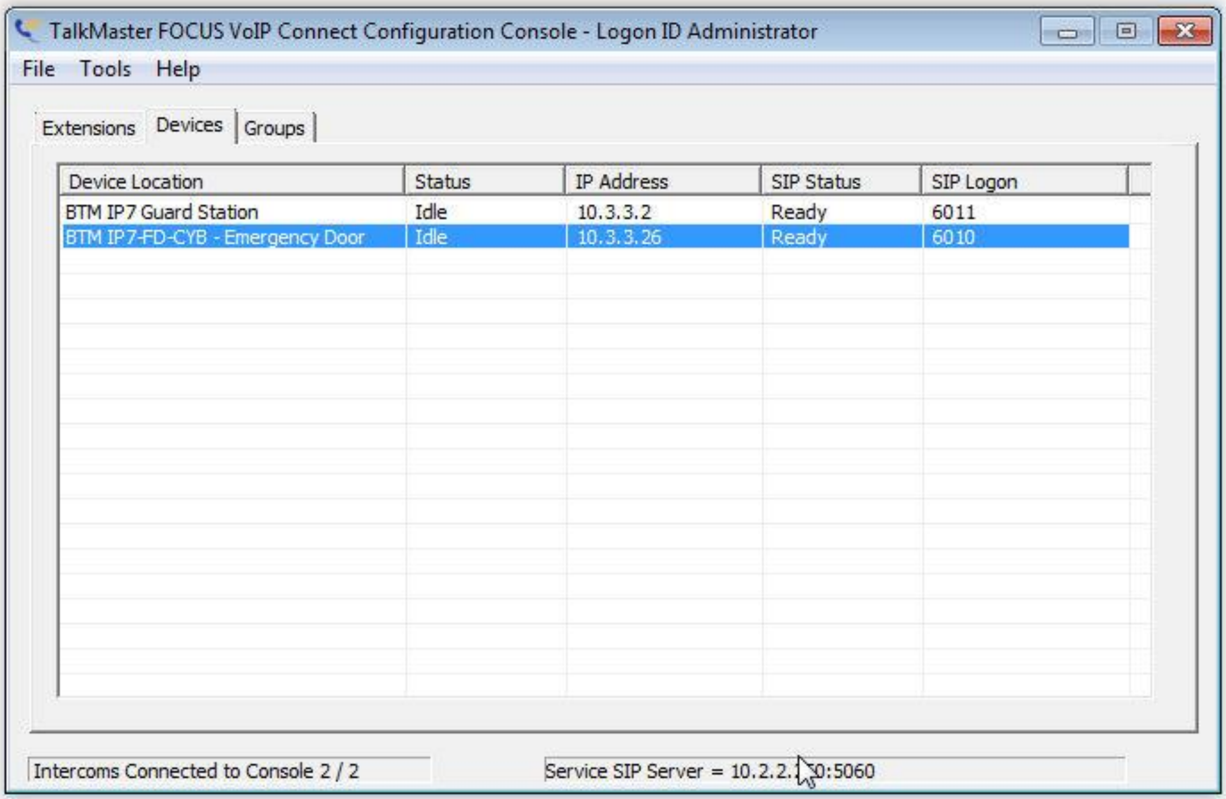
- **Extensions** tab - This tab shows all of the extensions that have been assigned to either IP Endpoints or to Paging Groups
- **Devices** tab - This tab shows all of the IP Endpoints that have been defined in the Admin Console and any extensions that have been assigned to them
- **Groups** tab - This tab shows all of the **Paging Groups** that have been defined in the Admin Console and any extensions that have been assigned to them

Column descriptions:

- **Name** - Displays the name of the IP Endpoint or Paging Group that has been assigned an extension
- **SIP Status** - Displays the SIP status of the IP Endpoint or Paging Group
- **Logon ID** - Displays the extension that is assigned to the IP Endpoint or Paging Group
- **Intercoms Connected** - Displays the number of IP Endpoints defined in the Admin Console and the number of them that are currently connected to the **TalkMaster FOCUS Server** service
- **Service SIP Server** - Displays the address of the SIP Server that the **TalkMaster FOCUS VoIP Connect Servic**e is communicating with

Devices tab

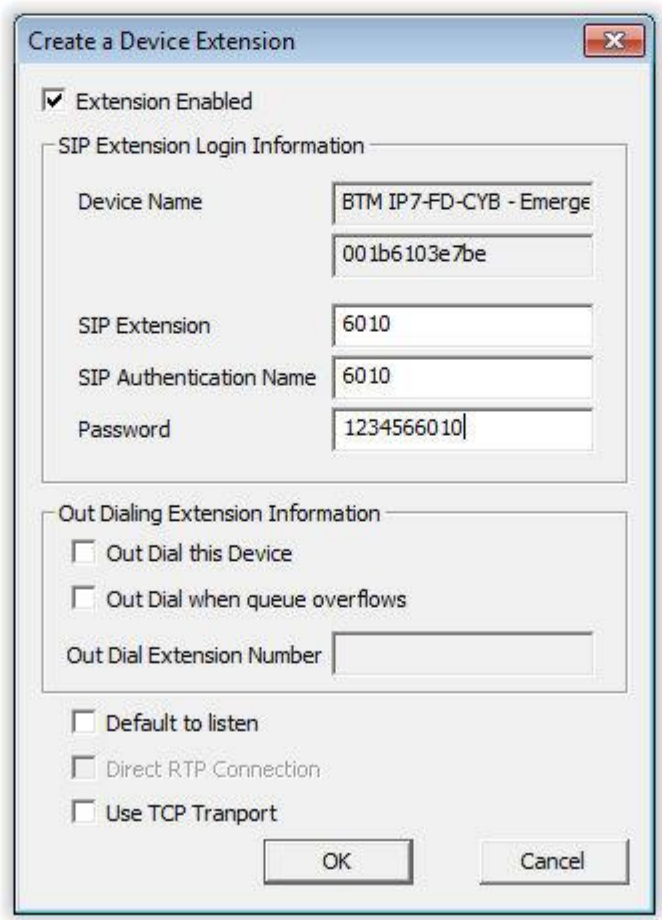
The devices tab shows all of the IP Endpoints that are configured to connect to the TalkMaster Focus Server, their connection and SIP statuses.



- **Device Location** - The Location Name of the IP Endpoint that has been configured in the Admin Console
- **Status** - The current status of the IP Endpoint in TalkMaster FOCUS Server
- **IP Address** - The current IP Address of the IP Endpoint
- **SIP Status** - The current SIP Status of the IP Endpoint in SIP Server
- **SIP Logon** - The extension in the SIP Server that has been assigned to this IP Endpoint

Device Extensions

Device extension creation and editing is performed by **double clicking** on a device in a row.



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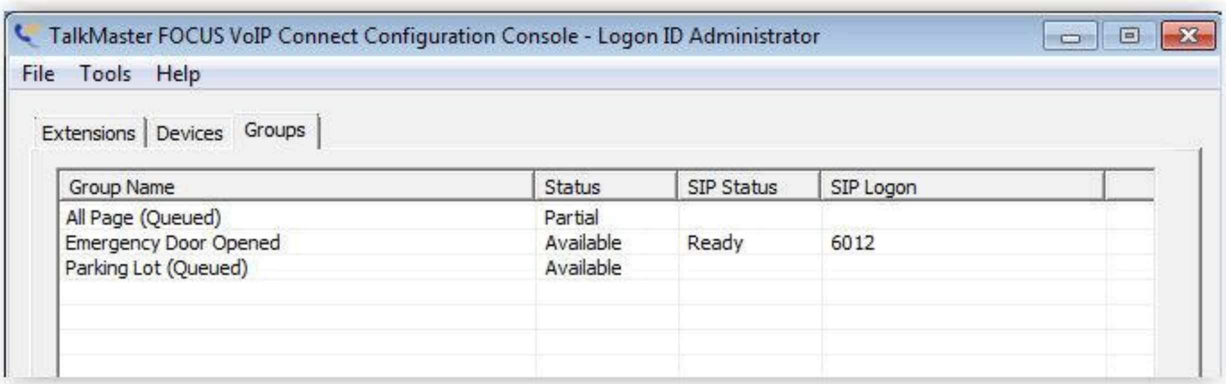
- **Extension Enabled -** To enable a SIP extension for a device, check this box. Then the SIP Extension, SIP Authentication Name, and Password are enabled and can be entered
- **Device Name** - Displays the name of the IP Endpoint that has been defined in the Admin Console
- **MAC Address** - Displays the network MAC Address of the IP Endpoint. The last six digits are the ICOM ID
- **SIP Extension** - The extension that is being assigned to this **IP Endpoint** and that has been defined in the SIP based phone system
- **SIP Authentication Name** - Some SIP based phone systems associate an Authenticated User ID with an extension and require it to be specified when logging on to the SIP based phone system. If an Authentication ID is not required by the SIP based phone system, then enter the extension. Some systems refer to this field as the **SIP Authorization Name**
- **Password** - The password that is assigned to the extension in the SIP based phone system. Some systems refer to this as the **Secret**
- **Out Dialing Extension Information**
    - **Out Dial this Device** - Check this box to have the IP Endpoint dial an extension or phone number when it's **TALK** switch is engaged

  - **Out Dial when queue overflows** - Check this box to have the IP Endpoint dial the extension or phone number when it's **TALK** switch is engaged AND when the TalkMaster FOCUS Server **Queue** it is in goes into Overflow status (as defined in the Admin Console). A **Queue** can go into overflow status based on:
    - The amount of time that has passed since the IP Endpoint's **TALK** switch has been engaged
    - The current number of unanswered calls in the **Queue**
    - No **TalkMaster FOCUS Server** Operator is available to answer the call
  - **Out Dial Extension Number** - Enter an extension or a phone number (including any digits required to get an outside line) to have the IP Endpoint dial
- **Default to Listen** - By default, a half-duplex call will be playing audio from the IP Endpoint to the call source. If the default action should be to record audio from the IP Endpoint, check this option
- **Direct RTP Connection** - If the RTP connection should be directly between the SIP Server and this IP Endpoint, check this option. NOTE: This requires that the IP Endpoint is accessible and routable from the other endpoint
- U**se TCP Transport** - The VoIP Connect will use the UDP Transport by default when creating SIP packets. Check this box to use the TCP Protocol for SIP packets for this extension

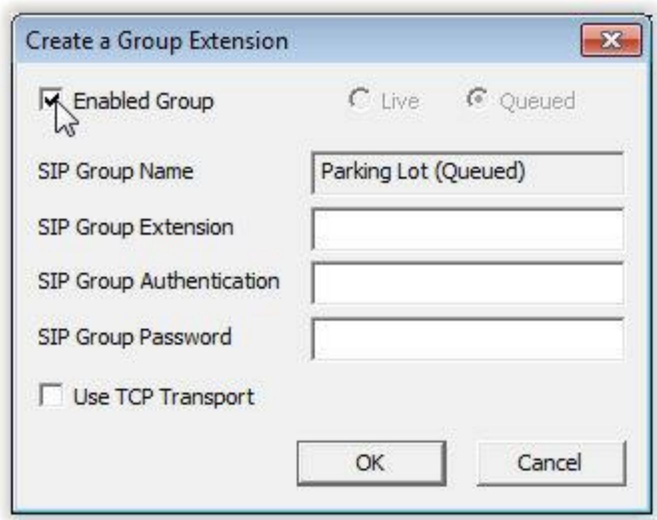
Groups tab

VoIP Connect can assign an extension defined on a SIP base phone system to a group of IP Endpoints. This provides the ability to dial the extension and make an audio page to the IP Endpoints. The **Groups** tab shows all of the Paging Groups, Status, and SIP status.**TalkMaster FOCUS Server** based



Group Extensions

Group extension creation and editing is performed by **double clicking** on a row for the group.



- **Enabled Group -** In order to edit this group, and for it to begin registering and be in use, the **Enabled Group** checkbox must be checked
- **Live** or **Queued** radio button - Server based Paging Groups are defined in the Admin Console as either **Live** or **Queued**. When an audio page is sent to a **Live** group, the audio is sent and played on the IP Endpoints while the operator is talking on the phone. There will be a 1 to 2 second delay from the time the audio is spoken till it is played on the IP Endpoints. For instances when the phone operator is in or near the speakers, a **Queued** group can be used. For Queued groups, the phone operator's audio page is recorded and then sent to the IP Endpoints when the operator hangs up the phone
- **SIP Group Extension** - The extension that is being assigned to this **Paging Group** and that has been defined in the SIP based phone system
- **SIP Group Authentication** - Some SIP based phone systems associate an Authenticated User ID with an extension and require it to be specified when logging on to the SIP based phone system. If an Authenication ID is not required by the SIP based phone system, then enter the extension
- **SIP Group Password** - The password that is assigned to the extension in the SIP based phone system
- **Use TCP Transport** - The VoIP Connect will use the UDP Transport by default when creating SIP packets. Check this box to use the TCP Protocol for SIP packets for this extension

OK - Saves the information and exits

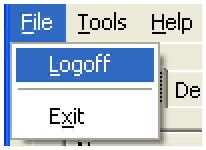
Cancel - Ignores any changes and exits

Menu Options

File Menu

File->Logoff - Select this option to disconnect from the TalkMaster FOCUS Server without exiting the console application

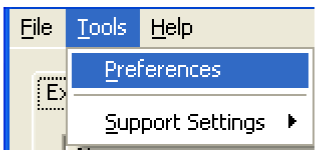
File->Exit|- Select this option to disconnect from the TalkMaster FOCUS Server and exit the console application.



Tools Menu

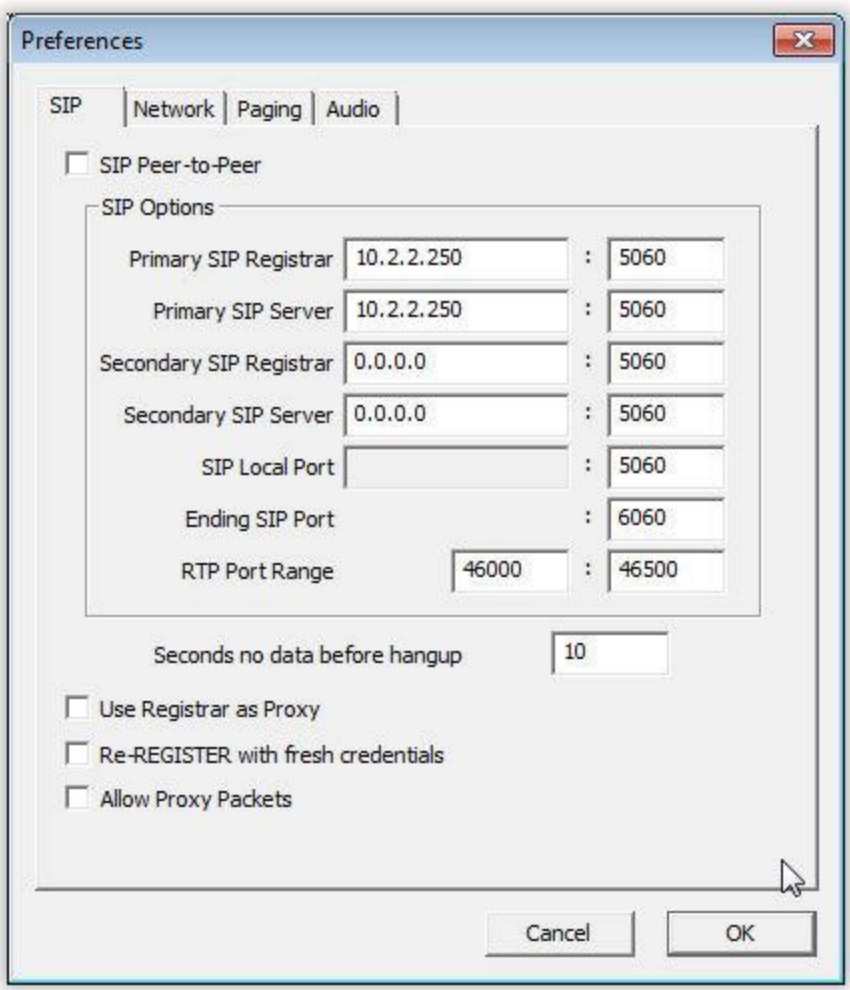
Tools->Preferences - Opens a four tabbed dialog used to configure settings that the **VoIP Connect Service** used to communicate with the SIP Based phone system.

Tools -> Support Settings - These settings should only be set when instructed by Digital Acoustics Support



SIP Tab

The SIP tab is used to configure the settings that the **VoIP Connect Service** will use to communicate with the SIP based phone system



- **SIP Peer-to-Peer** - Check this option to enable Peer-to-Peer mode. In Peer-to-Peer Mode, no Primary/Secondary SIP Server will be used. Calls to and from IP Endpoints will need to specify the IP Address along with the extension of the device being called. To call extension 6110 located at IP Address 10.2.2.110, the dialing extension would be specified as 6110@10.2.2.110
- **T**he **Primary SIP Server** (also known as the **Proxy Address**) is typically the same as the **Primary SIP Registrar**. If this is different than the **Primary Registrar**, then the **Use Registrar as Proxy** box must be checked
- **Secondary SIP Server /Registrar** - VoIP Connect Service supports a Secondary SIP Server. If communication is lost with the Primary SIP Server, the VoIP Connect Server will attempt to register all of the defined extensions with the Secondary SIP server. If the Primary SIP Server becomes available again, VoIP Connect Service will re-register all of the devices with the Primary SIP Server
- **SIP Local / Ending Ports** - Since each IP Endpoint / Paging Group will be registering with the IP Address of the VoIP Connect Service, each will need it's own "port". Make sure to allocate at least as many ports as extensions that will be registered
- **RTP Port Range** - Each SIP VoIP session (INVITE) will have a different RTP port pair. Specify at least as two times as many ports as extensions
- **Seconds no data before hangup** - This specifies how many seconds to wait between RTP packets before the VoIP Connect Service considers a call "timed out" and aborts the call

- **Use Registrar as Proxy** - Some external SIP services expect the Server information to always be sent to a proxy address, rather than the Server address.
- **Re-register with fresh credentials** - Some SIP server do not want the previously used credentials to be re-used. Instead, they only want unused credentials when the REGISTER packet is sent
- **Allow Proxy Packets** - This option specifies whether SIP Proxy Services are going to be sending packets from an address that is not the Server or the Registrar address.

Cancel - Select this option to ignore any changes to the Preferences and exit to the main screen

OK - Select this option to accept any changes to the Preferences and exit to the main screen

Network tab

The **Network** tab is used to configure some advance options

External IP Addresss Info

If a SIP Application Layer Gateway is available, then use that **instead** of changing any of these options.

When a NAT'd firewall is betwen the VoIP Connect Service and the SIP based phone system, the external IP Address of the VoIP Connect Service must be passed in the SIP packets instead of the VoIP Connects local IP Address.

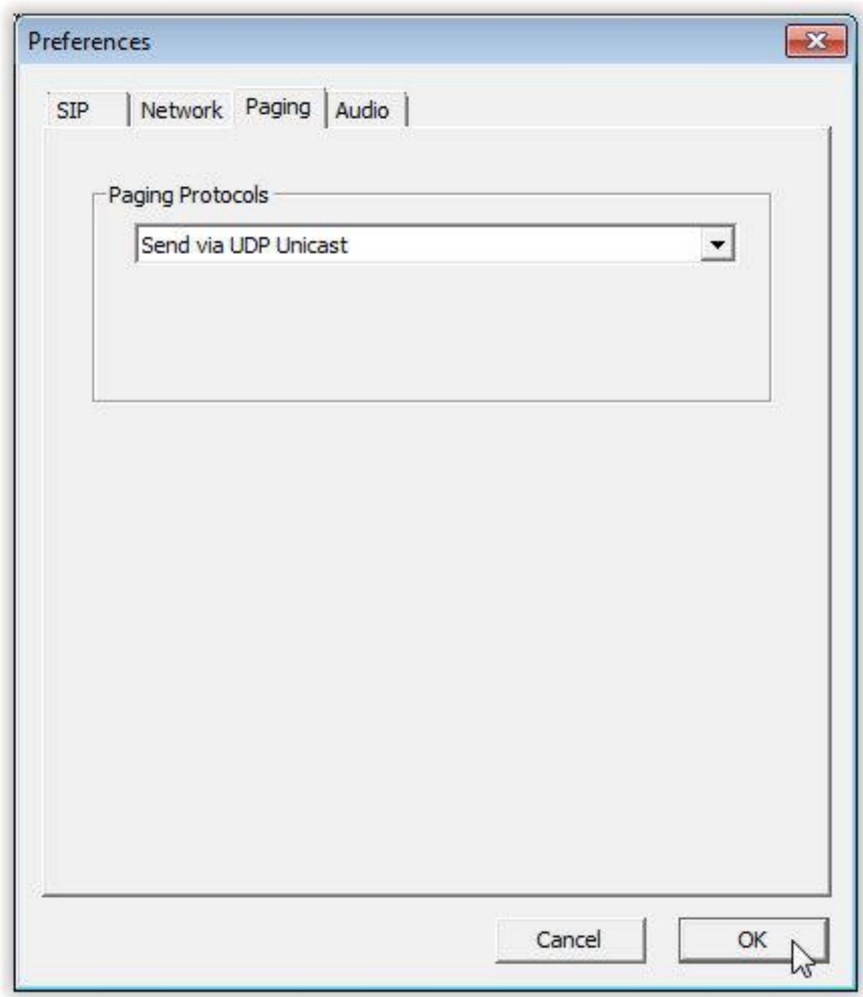
- **External IP Address** - Check this box and either enter the external IP Address of the VoIP Connect Service or enable to STUN option to automatically determine it
- **Use STUN** - Check this option to automatically determine the External IP Address of the VoIP Connect Service the next time the VoIP Connect Service is restarted. The results of the STUN testing will appear below Firewall Analysis. Some Firewalls will prevent this from working.
- **Non-Direct RTP Packets Between** - If the audio path is running through the VoIP Connect Service (non-direct), the RTP Packet size can be specified to be between the two values shown, with a default “offer” of a number between those two values.

Cancel - Select this option to ignore any changes to the Preferences and exit to the main screen

OK - Select this option to accept any changes to the Preferences and exit to the main screen

Paging tab

The **Paging Protocols** tab is used to override how the **TalkMaster FOCUS Server** service sends audio to **Paging Groups**. Typically, this should use be set the same as the **Admin Console --> Setup --> Paging Options** tab



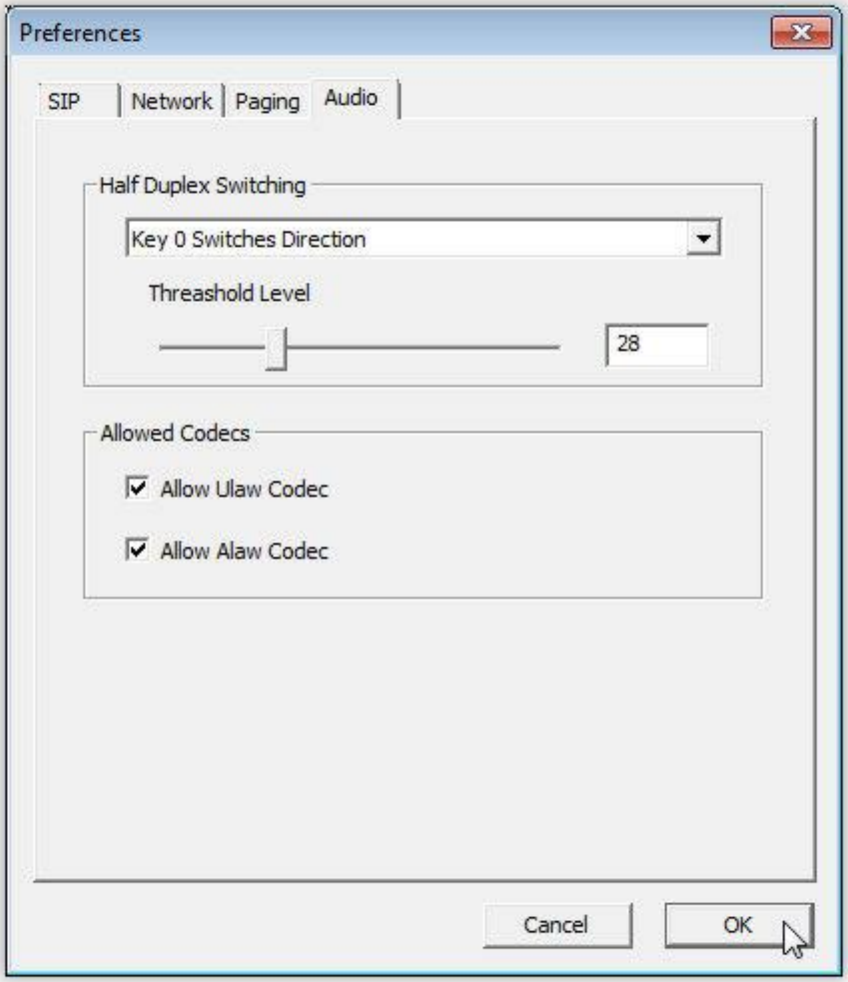
Audio tab

This tab is used to setup audio options for SIP IP Endpoints.

The **Half Duplex Switching** option determines how the audio direction is to be changed when communicating with Half Duplex IP Endpoints:

- **Key 0 switches Direction -** When this option is selected, the phone operator presses the 0 (zero) key on the phone to switch between Talking and Listening with an IP Endpoint
- **VAD / Automatic** - When this option is selected, the audio level from the phone is monitored by the VoIP Connect Service and when it falls below the Threshold Level, the system will start sending audio from the IP Endpoint to the phone. When the audio level rises above the Threshold Level. the system will start sending audio from the phone to the IP Endpoint. Please note, that use of this setting requires a very quite environment around that phone and that it can take up to two seconds before the VoIP Connect Service gets enough data to determine when to switch the audio direction
  - **Threshold Level** - Specify the relative audio level received from the phone that the VoIP Connect Service will use to switch the audio direction

- **Allowed Codecs** - The IP Endpoints support the uLAW and aLAW codecs Specify at least one of them and make sure it is available on the SIP based phone system



Support

Contacting Technical Support

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