MicroComm DXL

DXL VoIP Topics

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1   Introduction

This application note provides basic information on the DXL Voice Over IP protocol (VoIP) when used in a DXL VoIP system.

You should have some knowledge of the DXL Administrator Software and networking before using this guide.

2   DXL Voice Over IP Basics

2.1 DXL Networking Overview

The DXL system allows you to use analog master stations, intercom stations, and paging along with the Voice Over IP devices. This document only shows networking required for the VoIP connected devices.

A DXL system using Voice Over IP trunking (VoIP trunking) uses standard Ethernet networks using Voice Over IP protocols to transport audio throughout the system.

There are three types of networking packets transmitted throughout the system:

1. Inter exchange call setup: When a connection is required between exchanges (DCCs), the DCC initiating the connection will set up a VoIP audio trunk will use the DXL networking protocol to indicate to the other DCC(s) that it wants them to connect up the VoIP audio trunk to the end device located on that other DCC. The DXL networking protocol uses Harding specific network packets sent by unicast TCP to port 50099 on the destination DCC(s).

2. SIP setup: When a VoIP master or station is first powered up, it will initiate a SIP request to the DCC to tell the DCC that it wants to register its connection path with the DXL system. This will use the standard SIP (Session Initiation Protocol) using unicast UDP to the standard SIP server port on UDP port 5060 of the DCC.

3. RTP audio streams: The actual audio between devices will be passed as a standard RTP (Real Time Protocol) audio stream. The DXL system will always use the "PCMU" or "G.711 u-Law" payload type (uncompressed 8 kbit/sec bitstream). Each direction of the call will establish a one-way stream. Thus a connection from a master station to a DCC would have two components - one initiated by the master which will connect the master's microphone output to the DCC's microphone input, and the other initiated by the DCC which will connect the DCC's speaker output to the master's speaker input).

There are three different types of devices that will use the RTP audio streams.

a. Exchange (DCC) VoIP trunk:

The Exchange VoIP trunking allows voice to be transmitted from one DCC to any other DCC(s) in the system. This allows a master connected to one DCC to communicate to a station or other device on another DCC. Whenever a connection between one DCC to another is required, the DCC will open up an RTP stream and will constantly distribute the audio to any DCC in the system that needs to receive the audio.

DXL systems with firmware 5.1.1 and lower will always use standard RTP protocol using **broadcast** UDP packets using one of the ports between 16394 and 16470.

DXL systems with firmware 6.0.0 and higher will use standard RTP protocol, configurable either for **broadcast** or **multicast** UDP packets, and also can configure the starting RTP port (16512 is the default). If using multicast, your network administrator will need to assign a multicast address   and set a TTL (Time to Live) value for the DXL audio communication. See section 3.3 "Using Multicast VoIP Exchange Trunk in the DXL System" for details.

If this connection involves a two-way talk path, the DCC on the other side will also open up an RTP stream, again using the same RTP protocol as above. These connections are only active when an inter exchange call is in progress. They will be closed when the inter exchange call is disconnected. **Note: Your network administrator will have to take special care that these broadcast or multicast packets will be received by all DCCs in the system.**

b. Master VoIP audio:

Each master or VoIP phone in the system, when powered up, will initiate a connection to one specific DCC (the address specified in the master's settings). This connection will use an audio stream using the standard RTP protocol using unicast UDP packets and will be addressed to the IP address of the DCC. The port number range is dependant on the configuration settings of the master and will increment with each connection (the default may vary between different models of master or phone but many Harding masters will use 16384 for the default starting address). The DCC will also open the reverse channel, a RTP stream using unicast UDP packets addressed to the IP address of the master. For Harding masters (IMS-640 and TMM-640) this connection will be always active (there will be an active UDP stream in both directions at all times) – this allows the DXL system to sound call queued up tones without having to cause the master to ring. For third party VoIP devices such as IP phones or PDA devices, this connection will only be active when there is an audio connection active to that device (so it will be active only when the master is in a call).

Note: The RTP stream the master uses should not conflict with the range that the DCC uses for the Exchange VoIP trunks. With older DXL firmware versions which start at 16394, you should avoid starting ports anywhere in the 16394 to 16470 range. This means that for older DXL firmware versions, on many master stations you will need to change the default RTP starting port on the master station.

    c.   Station or Paging Speaker VoIP audio:

If you have any VoIP intercom stations or paging speakers, they will also use an RTP audio stream. In this case, the station or speaker will be called from any DCC on the system directly (or in cases even other by another station or speaker). When the station or speaker has an audio connection active (when it is in a call) it will open a RTP stream using unicast UDP to the DCC (or other device) it is connected to. Likewise the DCC (or other device) will open up a RTP stream using unicast UDP to the station. The starting port number of the UDP stream is also configurable with the default usually being set to 16384, and will increment with each connection.

Note: The RTP stream the station or speaker uses should not conflict with the range that the DCC uses for the Exchange VoIP trunks. With older DXL firmware versions which start at 16394, you should avoid starting ports anywhere in the 16394 to 16470 range. This means that for older DXL firmware versions, on stations you will need to change the default RTP starting port on the station.
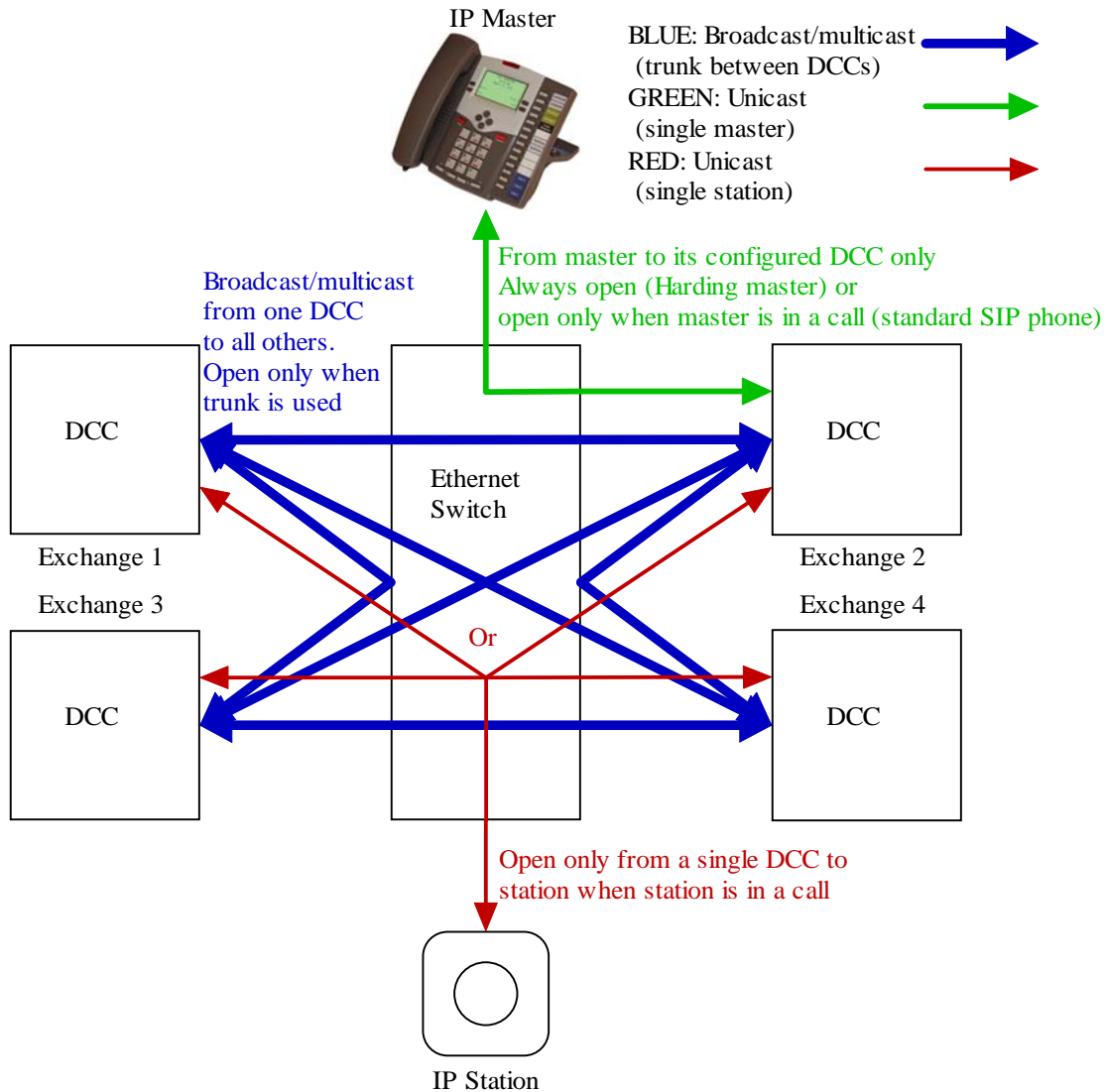
In summary, audio streams :

- Use RTP protocol
- Use UDP packets
- Use Unicast packets for masters and stations connecting to/from a DCC
- Use Unicast, Broadcast, or Multicast packets for exchange VoIP trunk with DCCs connecting to/from other DCCs
- **Note: Use of Broadcast or Multicast packets means that your network equipment is required to pass these messages throughout the DXL network, so the network will have to be configured as one of the following methods:**
  - All DXL systems need to be on one subnet with the routers/switches passing all broadcast messages
  - All DXL systems need to be on a VLAN with the routers/switches passing all broadcast messages
  - All network routers/switches with DXL systems on them need to be configured to pass all broadcast messages between at least these routers/switches.
  - Use multicast and set the multicast address to a locally routed address
  - Use multicast and set a multicast address to an Ad-Hoc or administratively-scoped range, and configure all network routers/switches to forward these multicast messages to devices that subscribe to this address.

    Some of these methods will be described in detail in below sections.

Drawings showing the DXL networks are shown below.

IP Master

BLUE: Broadcast/multicast
 (trunk between DCCs)
GREEN: Unicast
 (single master)
RED: Unicast
 (single station)

From master to its configured DCC only
Always open (Harding master) or
open only when master is in a call (standard SIP phone)

Broadcast/multicast
from one DCC
to all others.
Open only when
trunk is used

DCC

Exchange 1

Exchange 3

DCC

Ethernet
Switch

Or

DCC

Exchange 2

Exchange 4

DCC

Open only from a single DCC to
station when station is in a call

IP Station

2.2 DXL network connection from VoIP master to standard intercom

The following series of drawings show the steps involved in a VoIP DXL system when using VoIP masters and standard intercoms.

1. When the DXL system starts or a master is turned on, each master registers with the DXL DCC using SIP, then establishes a two-way RTP audio stream with the DCC. This two way RTP stream is permanent until the master or DCC is turned off or rebooted, and is re-established when both devices are back online.

   a. The master will use a UDP SIP request to register with one specific DCC that is configured in the master.

   b. The DCC will open a unicast UDP RTP audio stream to the master, and instruct the master to open a unicast UDP RTP audio stream back to the DCC.

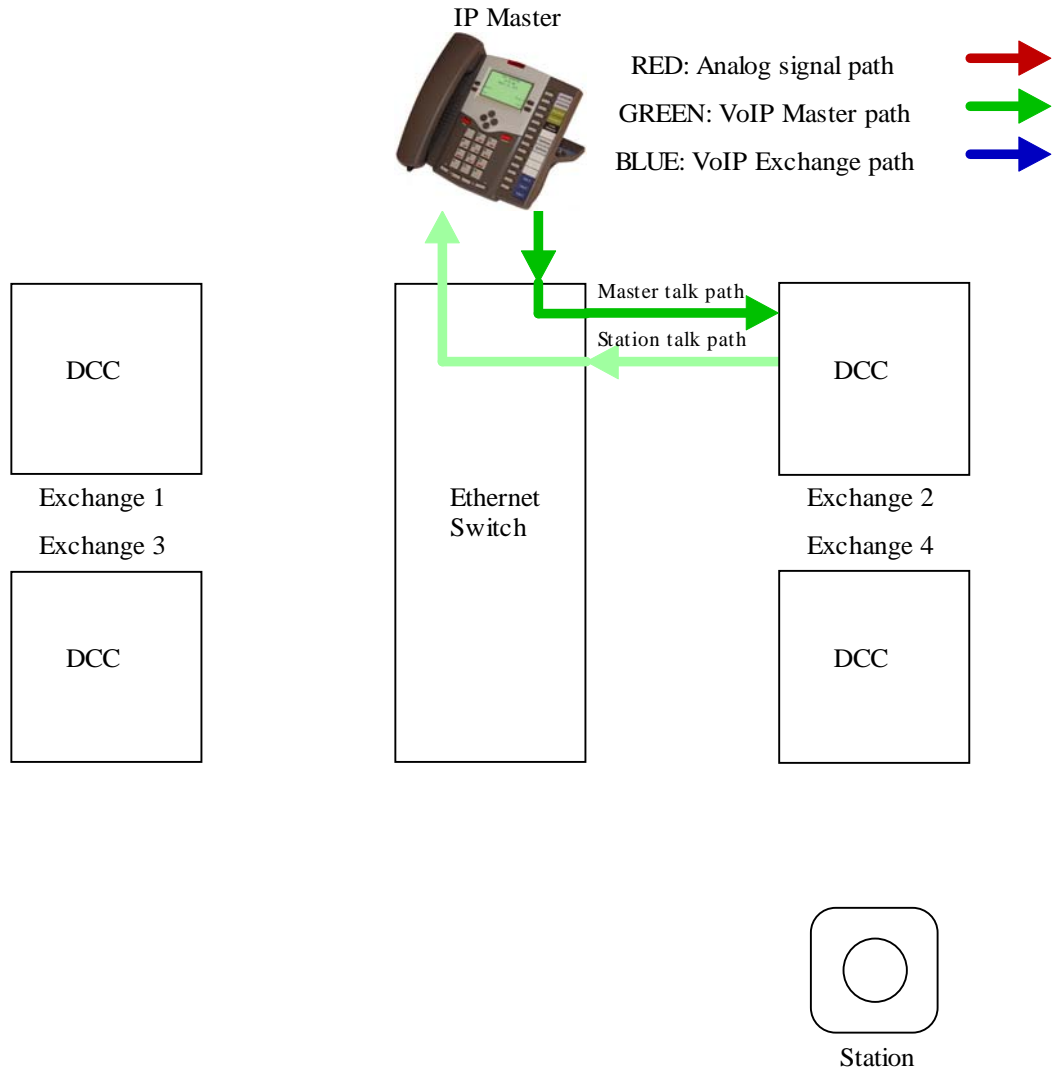The resulting master VoIP audio streams are shown in green below.

IP Master

RED: Analog signal path

GREEN: VoIP Master path

BLUE: VoIP Exchange path

DCC

Exchange 1

Exchange 3

DCC

Ethernet Switch

Master talk path

Station talk path

DCC

Exchange 2

Exchange 4

DCC

Station

Figure 1: Master VoIP streams initiated on startup

2. Whenever a connection is established from a master station (VoIP or standard) to an intercom on another exchange, broadcast or multicast RTP audio streams will be sent between exchanges to establish the voice connection.

   a. If using firmware earlier than version 6.0.0, the master's DCC will open up a broadcast or multicast UDP RTP stream to all DCCs (including the DCC with the station) and instruct the station's DCC to listen to this stream through the DXL networking protocol on port 50099. In turn, the station's DCC will open up a broadcast or multicast UDP RTP stream to all DCCs (including the DCC with the master) and the master's DCC will listen to this stream.

If using firmware 6.0.0 or later, the master's DCC will open up a unicast UDP RTP stream to just the station's DCC and instruct the station's DCC to listen to this stream through the DXL networking protocol on port 50099. In turn, the station's DCC will open up a unicast UDP RTP stream to the master's DCC and the master's DCC will listen to this stream.

b. The station's DCC will open an analog connection to the station.

The resulting exchange VoIP audio streams are shown in blue below, with the analog station connection shown in red below. Note that if using firmware 6.0.0 there will only be the connection from the master's DCC to the station's DCC (the paths marked "not used") will not be established since the connection will be unicast).
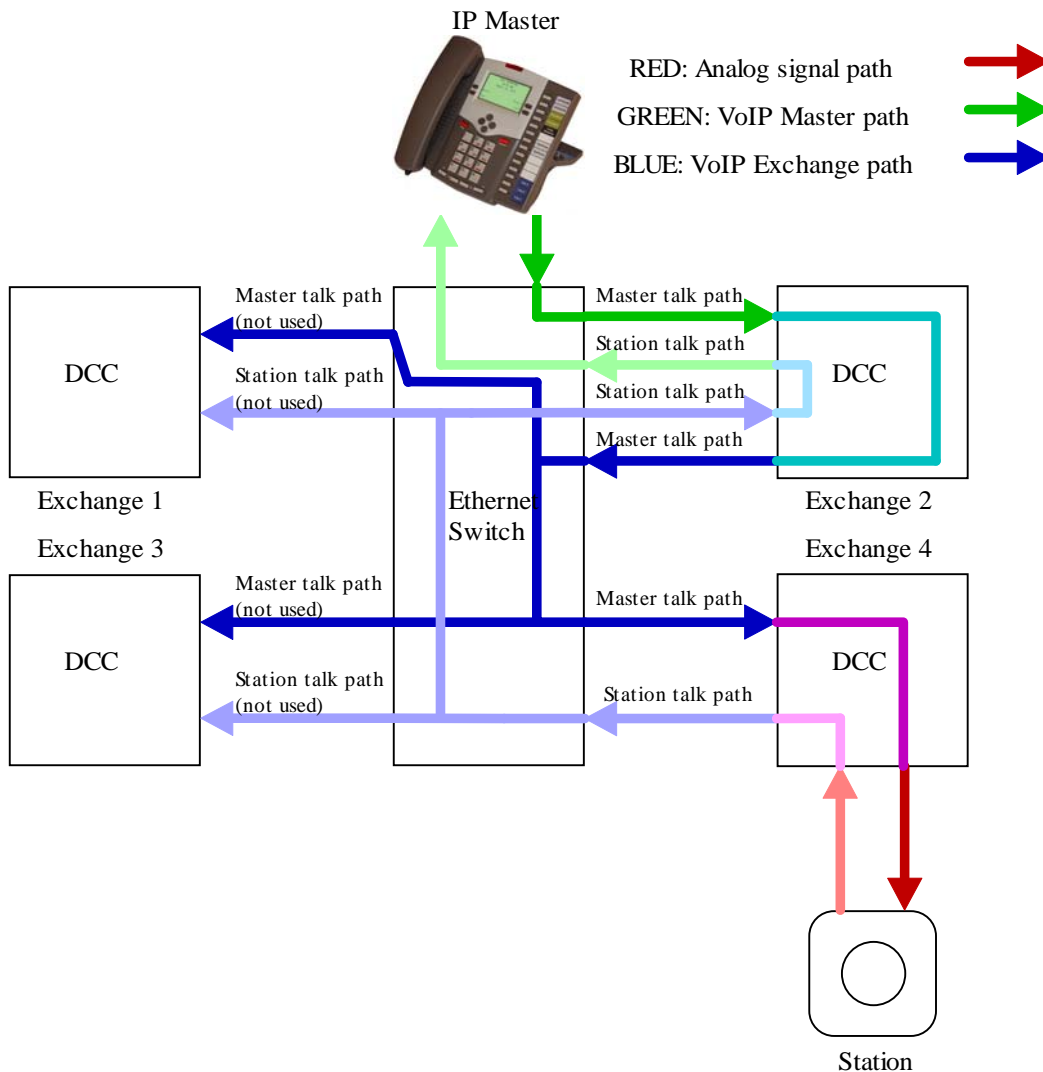

Figure 2: Exchange VoIP streams initiated when connected to a station

2.3 DXL network Connection from VoIP master to VoIP intercom

The following series of drawings show the steps involved in a VoIP DXL system when using VoIP masters and standard intercoms.

1. When the DXL system starts or a master is turned on, each master registers with the DXL DCC using SIP, then establishes a two-way RTP audio stream with the DCC. This two way RTP stream is permanent until the master or DCC is turned off or rebooted, and is re-established when both devices are back online.

    a. The master will use a UDP SIP request to register with one specific DCC that is configured in the master.

    b. The DCC will open a unicast UDP RTP audio stream to the master, and instruct the master to open a unicast UDP RTP audio stream back to the DCC.

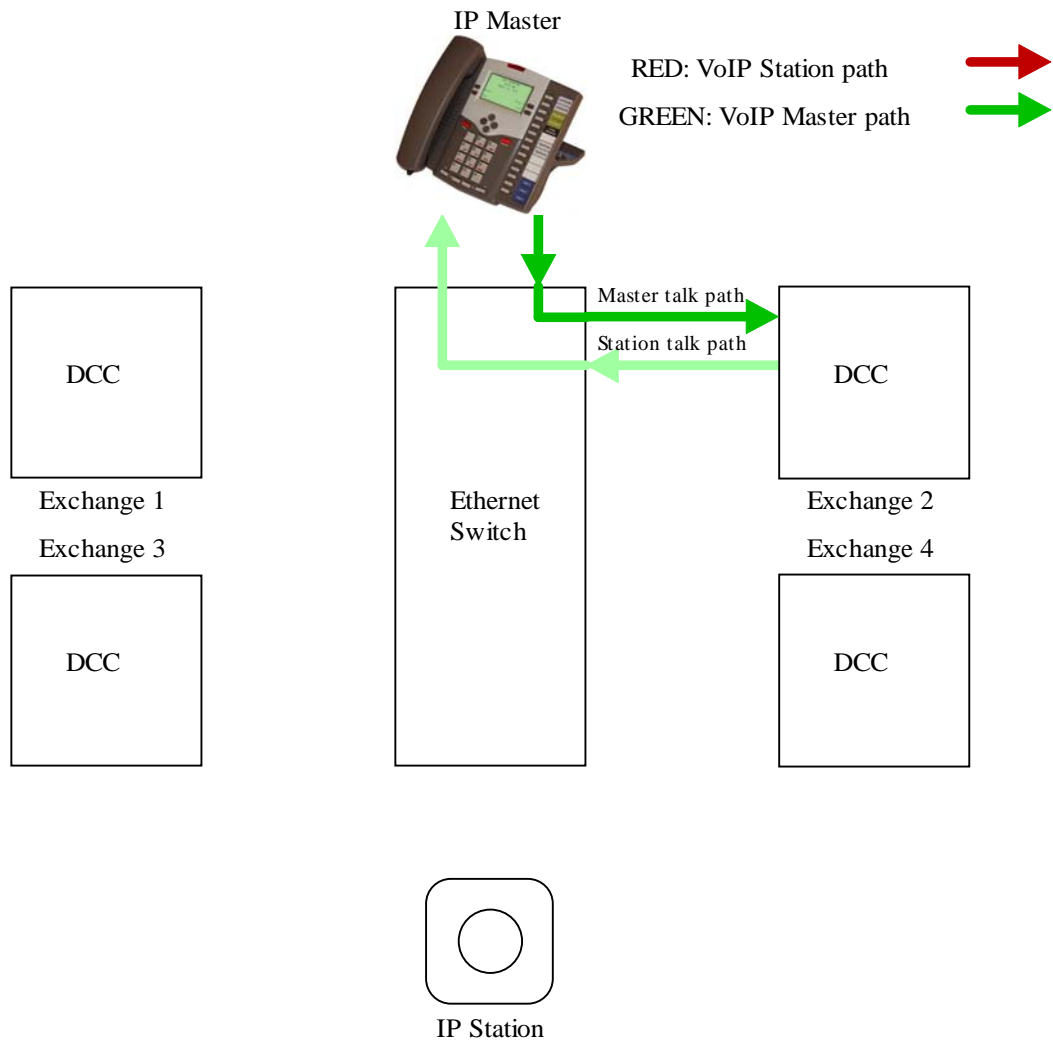The resulting master VoIP audio streams are shown in green below.

Figure 3: Master VoIP streams initiated on startup

2. Whenever a connection is established from a master station (VoIP or standard) to a VoIP intercom, unicast RTP audio streams will be sent directly to the station to establish the voice connection.

   a. The master's DCC will send a command to the station through the DXL networking protocol. This command instructs the station to listen to the DCC's audio stream and in turn send its audio stream to the master's DCC.

   a. The master's DCC will open up a unicast UDP RTP stream to the station and listen to the unicast UDP broadcast RTP stream from the station, and in response to the above command, the station will open up a unicast UDP RTP stream to the master's DCC and listen to the unicast UDP RTP stream from the master's DCC.

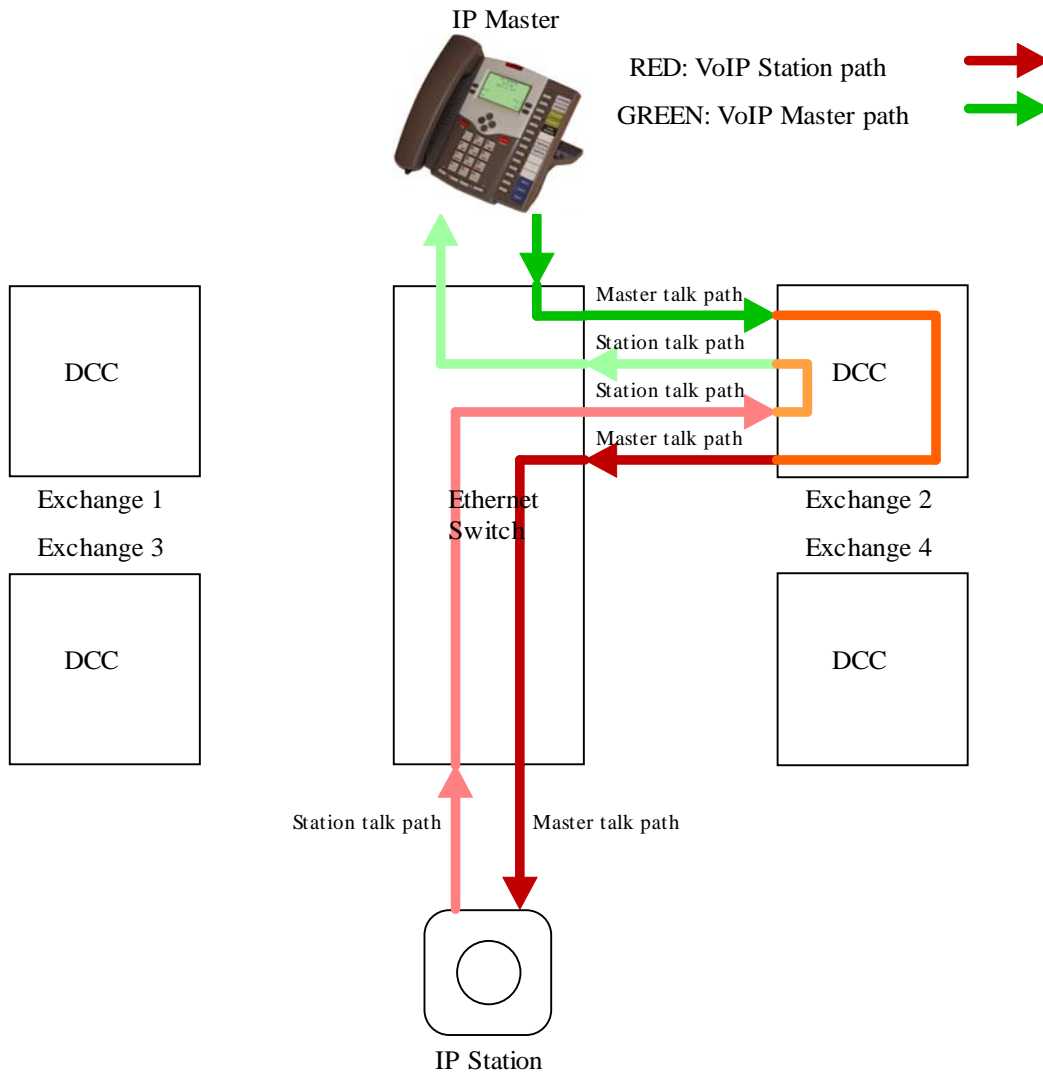The resulting exchange VoIP station audio streams are shown in red below.

Figure 4: Exchange VoIP streams initiated when connected to a station

Not shown in the above: When a VoIP station is powered up, it sends a UDP SIP registration packet to the DXL system to announce that it is available to be connected to. No audio connections are made at that time.

3 Quality of Service (QoS)

3.1 Basics of Quality of Service

Quality of Service (abbreviated QoS) is a method of prioritizing certain network packets such that if the network is very busy, these packets have priority and will be sent first. This allows data that needs to be sent real time (such as audio) to be prioritized so that if the network is busy, the important packets are received on time (in this case, audio packets are sent in time so that the audio is not broken up due to the next portion of the audio not being received before the last previous portion of the audio has finished being sent to the speaker).

If all network devices are connected to one network switch, QoS usually is not required as each device on the switch has unrestricted access to any other device on the switch. A simple network switch will not likely have the option to have QoS applied to it.

However, in a larger system, there will likely be a backbone network upon which all communications between switches take place. In this case, you may need QoS on the system so that if the backbone network is busy, the real time packets such as audio will be given priority over other network traffic. Usually backbone networking is done with more complicated routers/switches which will likely have QoS options.

QoS has two components:

The device that has real-time requirements has to indicate to the switch what the priority of its real time packets are. In the case of the DXL intercom system, the DCC's will indicate the priority the RTP (Real Time Protocol) audio packets through the front panel setting accessed from the "Setup", "Ethernet IP/Mask", "RTP DSCP" setting. This will indicate the priority level of these packets, from normal (same as everything else) to highest priority.

Networking equipment which has QoS functionality will have some method of determining what to do with the packet priorities.

There are two ways which networking equipment will handle priority numbers.

The older method is assigning a "TOS" priority. Newer methods use a "DSCP" priority level. DXL systems assign priorities based on the DSCP method. You can translate a TOS value to a DSCP value with the table that follows.

In general:

1. Make sure your networking equipment is capable of QoS prioritization.

2. Find or configure an appropriate priority level in your networking equipment that is appropriate for VoIP audio. For high priority VoIP packets, a common DSCP priority level to use is "ef" ("Expedited" setting on the DXL front panel).

3. Configure all DXL DCC's to assign that priority level for RTP packets.

Below is a table cross referencing TOS priority levels with DSCP priority levels, along with the DSCP class and DXL name for that class.

TOS (Dec)	TOS (Hex)	TOS Precedence (Dec)	TOS Precedence Name	TOS Delay flag	TOS Throughput flag	TOS Reliability flag	DSCP (Hex)	DSCP (Dec)	DSCP/PHB Class DXL name in ()
0	0x00	0	Routine	0	0	0	0x00	0	None (Default)
32	0x20	1	Priority	0	0	0	0x08	8	cs1 (Class 1)
40	0x28	1	Priority	0	1	0	0x0A	10	af11
48	0x30	1	Priority	1	0	0	0x0C	12	af12
56	0x38	1	Priority	1	1	0	0x0E	14	af13
64	0x40	2	Immediate	0	0	0	0x10	16	cs2 (Class 2)
72	0x48	2	Immediate	0	1	0	0x12	18	af21
80	0x50	2	Immediate	1	0	0	0x14	20	af22
88	0x58	2	Immediate	1	1	0	0x16	22	af23
96	0x60	3	Flash	0	0	0	0x18	24	cs3 (Class 3)
104	0x68	3	Flash	0	1	0	0x1A	26	af31
112	0x70	3	Flash	1	0	0	0x1C	28	af32
120	0x78	3	Flash	1	1	0	0x1E	30	af33
128	0x80	4	FlashOverride	0	0	0	0x20	32	cs4 (Class 4)
136	0x88	4	FlashOverride	0	1	0	0x22	34	af41
144	0x90	4	FlashOverride	1	0	0	0x24	36	af42
152	0x98	4	FlashOverride	1	1	0	0x26	38	af43
160	0xA0	5	Critical	0	0	0	0x28	40	cs5 (Class 5)
184	0xB8	5	Critical	1	1	0	0x2E	46	ef (Expedited)
192	0xC0	6	InterNetworkControl	0	0	0	0x30	48	cs6 (Class 6)
224	0xE0	7	NetworkControl	0	0	0	0x38	56	cs7 (Class 7)

Figure 5: DSCP and TOS Priority Levels

Some details about priority levels are as follows:

If dealing with TOS (Type of Service), the first 3 bits indicate the precedence, the 4th bit indicates the whether or not low delay is preferred, the 5th bit indicates whether or not high throughput is preferred, the 6th bit indicates whether or not high reliability is preferred and the 7th and 8th bits are reserved... More info can be found in RFC 791, written in 1981, which defines IP.

If dealing with DSCP (Differentiated Services, Diffserv) only the first 6 bits are used and the last 2 are ignored... More info can be found in RFC 2474, written in 1998, which defines the Differentiated Services Field (DS Field) which is what the TOS byte is referred to when talking about differentiated services and specifically DSCP. Also, RFC 2597 and RFC 3246 which define some of the PHB (Per-Hop Behaviour) classes may be useful reading.

3.2 Setting QoS Priorities in the DXL System

On the DXL system, you will need to set the priority level in each DCC on the system. Note that the initial (Default) firmware on the DXL system does not have this option. You need to upload and activate DXL Firmware 4.1.0b1 or above to be able to change this setting.

This setting is changed with the following sequence:

1. Press the "Setup" button on the front panel and go to the main "DCC Setup" menu if you are not there using the "Prev" key (or press "Setup" twice).

2. From the main "DCC Setup" menu, cursor down to the "Ethernet IP/Mask" line and press "Enter/Ack".

3. From the "Ethernet IP/Mask" menu, cursor down to the "RTP DSCP" line (which is not shown on the initial menus because it is an option below the "PPPL" setting).

4. On this line, press the right arrow to edit the setting.

5. Use up/down keys to change the setting to your desired setting (likely "Expedited"), then press "Enter/Ack".

6. Once in the Ethernet IP/Mask menu, press the "Prev" key. The front panel will prompt to restart for the changes to take place. Press "Enter/Ack" to reboot.
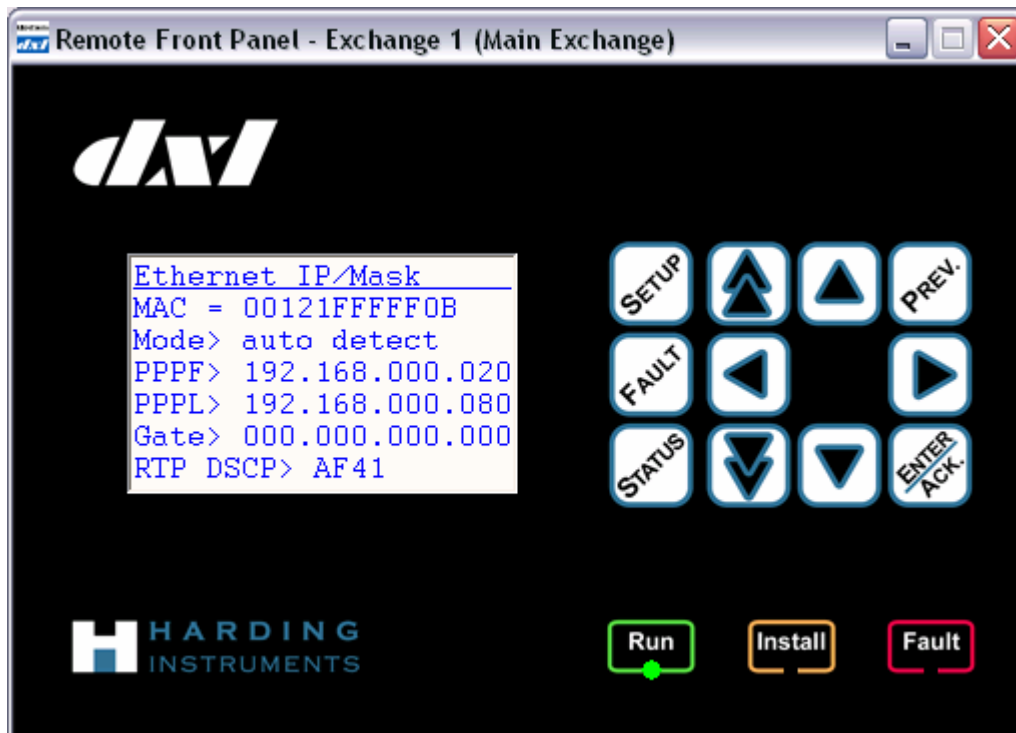

Figure 6: DXL Front Panel Showing RTP DSCP Setting

3.3 Using Multicast VoIP Exchange Trunk in the DXL System

DXL Firmware 5.0.0 in conjunction with DXL Administrator 1.5.0 introduced the ability to use multicast RTP packets for the Exchange VoIP trunk. In order to make use of this, your system administrator will have to assign a multicast address for this Exchange VoIP trunk and set up the network equipment appropriately, and you will need to set the multicast settings in the DXL Administrator.

The DXL system can be configured for IGMP snooping (using IGMP version 2) for switches that support IGMP.

The settings required are

1. Multicast IP address:

    a. If your switch or router does not have configuration options for multicast or IGMP, you must select a multicast address which is forwarded within the local network only. One such address range would be in the Local Network Control Block (224.0.0.0 to 224.0.0.255). Some of these are reserved for router control messages so you need to ensure that your network will not use the address you select. The range of 224.0.0.112 to 224.0.0.250 is usually unassigned, so you can likely use an address in this range such as 224.0.0.250. This will probably not require configuration in your switch or router. **Note that you will need to set this value manually as the default is set to 224.0.5.129 for the AD-HOC block below.**

    b. If your switch or router uses IGMP, you will need to assign an address in the DXL system which is normally not forwarded, then configure your router so that this address will use IGMP snooping. One such address range would be in the AD-HOC Block I from 224.0.2.0 to 224.0.255.255. An address within the 224.0.5.0 to 224.0.5.255 might be typically used. The DXL will default to 224.0.5.129.

2. Multicast TTL

    The Multicast TTL (Time to Live) value defaults to a value of 1. This indicates when this packet should stop forwarding to other networks. The default setting of 1 indicates that it will stay only on the local network will not be forwarded to another network (I.E. the Internet) if it requires 1 network hop (it will not pass between networks starting at the first network router it gets to). If you have a large internal network that has routers within the network between DXL systems, you will need to set this TTL value higher appropriately.

These settings are set in the DXL Administrator, "Configure: System" settings.
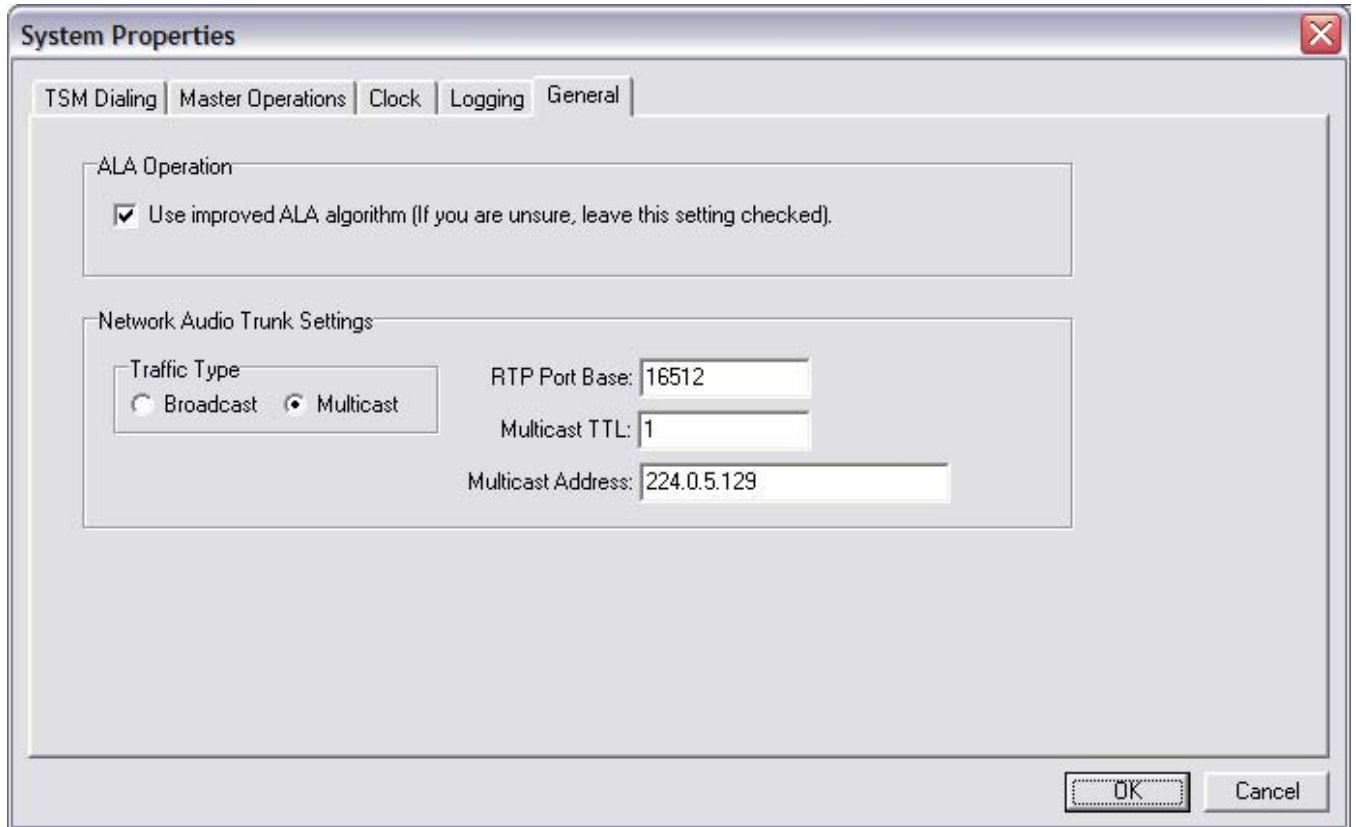
4   Network Requirements

4.1 Basic Network Requirements

Now that the basics of the DXL system are explained, the information your network administrator needs can be outlined.

4.1.1 Setting IP addresses and netmasks when DXL system is in a single subnet

If your network is going to be one subnet for the entire facility, then all of your network settings will be the same between all DCC's.

1. Netmask: Pick a network mask for your network. Many facilities have a /24 network with up to 254 devices, all with the first three octets of the network address the same, with the last octet denoting the device. In this case you would use a netmask of 255.255.255.0. Other network masks are possible

2. IP address: Each DCC will have a unique IP address, with the IP addresses of all DCC's being in the same subnet

3. Gateway address: If your facility has multiple networks which may need to exchange network packets (for example if your PLCs or server computers are on one network and the DXL systems are on another network) then you need to have a router or switch which will be connected to both networks. In this case you need to set the gateway address on each DCC to the IP address of the local router or switch that will

transfer packets to the other networks. In many cases there may be only one security network in which case you can leave the gateway address set to 0.0.0.0.

4. Set up the VoIP audio trunk:

   a. If you are using the default broadcast VoIP audio trunk, your network needs to allow the broadcast packets used by the VoIP exchange trunk to pass through the network. If you have simple network switches then this is usually the default. If you have a more complicated network, then you will need to allow broadcast packets between segments. If you do not want all network devices on your network to receive the broadcast packets, you can set up a VLAN (Virtual LAN) on your switches, and allocate the DXL DCC's, VoIP master stations, and VoIP intercom stations to this VLAN.

   b. If you are using the multicast VoIP audio trunk, you need to assign one Multicast IP address and Multicast TTL value and have your network switches set appropriately. See section 3.3 "

Using Multicast VoIP Exchange Trunk in the DXL System" above for details.

4.1.2 Setting IP addresses and netmasks when DXL system is in multiple subnets

If your network is going to have multiple subnets for different portions of the facility, then you will need to do the following:

1. Netmask: Pick a network mask for your network subnets. Many facilities have a /24 network with up to 254 devices, all with the first three octets of the network address the same, with the last octet denoting the device. In this case you would use a netmask of 255.255.255.0. Other network masks are possible

2. IP address: Each DCC will have a unique IP address. Different portions of the systems might be in different subnets however.

3. Gateway address: If you are using different subnets, each DCC must have its gateway address set to the router or switch responsible for connecting that subnet to the facility wide network.

4. Set up the VoIP audio trunk:

   a. If you are using the default broadcast VoIP audio trunk, your network needs to allow the broadcast packets used by the VoIP exchange trunk to pass through the network. However, on a network with routers in it, these packets are not passed through by default. You will need to set up your network to allow these broadcast packets to pass through. While you may be able to turn on a broadcast flag in your network equipment, this will result in the DXL Exchange VoIP broadcast traffic being sent to each networked device facility-wide. If you do not want all network devices on your network to receive the broadcast packets, you can set up a VLAN (Virtual LAN) on your switches, and allocate the DXL DCC's, VoIP master stations, and VoIP intercom stations to this VLAN. However, most facilities using multiple subnets would be best served by using multicast instead of broadcast, as below.

   b. If you are using the multicast VoIP audio trunk, you need to assign one Multicast IP address and Multicast TTL value and have your network switches set appropriately. See section 3.3 "

Using Multicast VoIP Exchange Trunk in the DXL System" above for details.