Tieline
The Codec Company

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IP – LINK QUALITY, DIAGNOSING FIREWALL RESTRICTIONS AND PACKET LOSS

PURPOSE

To Interpret Link Quality (LQ) and LQ Relationship to Firewall Blocking and understand the reasons for packet loss.

BACKGROUND

Tieline codecs connect over IP using TCP session connections which in turn establish the audio data path over UDP Ports. Tieline's built-in Link Quality diagnostics assist in determining problems affecting link and audio quality and possible blocks in firewalls, impeding throughput of audio. Tieline recommends using UDP over the audio path for remote broadcasts as it offers significantly higher link stability and lower audio delay than TCP.

IP LINK QUALITY

The IP link quality value consists of two digits. The value encodes two pieces of information. Refer to the table below.

- The Whole Number provides a general indication of the link quality over the last 8 seconds
- The Last Digit can be interpreted to give an indication of the worst reception event that has occurred in the last second.

Note: Ignore values for the first 10 seconds of connection

Whole Number	Cause	Effect on Audio
99	Perfect reception	Perfect Audio
96 - 98	Correction of data	Perfect Audio
85 - 95	Single packet late/loss	Good Audio almost undetectable artefacts to untrained ear
66 - 84	Multiple packet late/loss usually with FEC enabled	Imperfect Audio
55 - 65	Cyclic single packet late/loss	Reasonable Audio
44 - 54	Cyclic multiple packet late/loss	Poor Audio
33 - 43	Cyclic multiple packet late/loss	Bad Audio
22 - 32	Cyclic multiple packet late/loss	Terrible Audio
11 - 21	Cyclic continuous packet late/loss	Broken Audio (small patches of silence)
02 - 10	Drastic Loss	Broken Audio (large patches of silence)
01	Firewall Blocking (refer to Firewall Technical Note)	Session Data cannot be established - Call will not connect
00	100% Loss Empty Buffer	Silence

Last Digit	Event	Action	Effect on Audio
9	Perfect reception	None	Perfect Audio
8	Duplicate or disordered packets were received	Duplicates removed. Re-order	Perfect Audio
7	A single late/lost packet	Correction by FEC	Perfect Audio
6	Multiple late/lost packets	Multiple Corrections by FEC	Perfect Audio
5	A single late/lost packet	Single Repair	Almost undetectable to untrained ear
4	2 Consecutive or 2 to 4 non-consecutive late/lost packets	Multiple Repairs	Imperfect audio
3	3 to 4 consecutive late/lost packets	Multiple Repairs	Poor Audio
2	5 consecutive or 6+ non-consecutive late/lost packets	Multiple Repairs	Poor/Broken Audio
1	More than 5 consecutive late/lost packets	Multiple Repairs	Poor/Broken Audio (small patches of silence)

HOW TO DETERMINE WHERE FIREWALL PORT BLOCKING IS OCCURRING

If you find you are unable to either send or receive audio between the studio and remote codecs you can use Tieline's Link Quality reading to diagnose where ports are being blocked. LQ can be displayed on the front LCD screen of Tieline's Bridge-IT, Merlin and Genie codecs by selecting **Cxns**, then select the connection you want to view and press the **OK** button. LQ readings are also displayed on the home screen of all Commander and i-Mix G3 codecs.

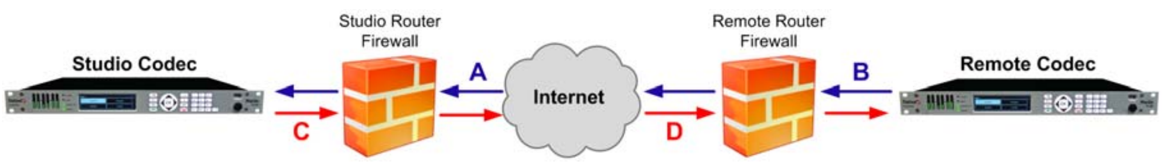
LINK QUALITY (LQ) READINGS

Send and Return LQ numbers help you to determine if a problem is occurring at either end of a connection. For example, on an IP connection the Return LQ reading represents the audio being downloaded from the network locally (i.e. audio data is being sent by the remote codec). Conversely, the Send LQ reading represents the audio data being sent by the local codec (i.e. being downloaded by the remote codec). To ensure a stable connection, try to maintain a reliable reading of 80 or higher for both the **Send** and **Return** LQ reading.

Important Note:
- The **Return** link quality reading is the same as the Local (**L**) setting displayed in early software versions on a G3 codec.
- The **Send** link quality reading is the same as the Remote (**R**) setting displayed in early software versions on a G3 codec.

Diagnosing Port Blocking via the Studio Codec LQ

If the studio codec **Return** LQ reading is **01** then incoming audio from the remote codec is being blocked by a firewall at either point A or B in the following diagram. If the studio codec **Send** LQ reading is **01** then outgoing audio from the studio is being blocked by a firewall at either point C or D in the following diagram.



Diagnosing Port Blocking via the Remote Codec LQ

If you attach your Tieline codec at the remote site to a LAN with access to the internet you can often dial and connect to the studio without any problem. It is less likely that a firewall will block outgoing TCP and UDP ports. However, if there is a firewall at the remote site it may block incoming data packets from the studio.

The principle is the same at the remote codec for diagnosing blocked ports. If the remote codec **Return** LQ reading is 01 then incoming audio from the studio codec is being blocked by a firewall at either point C or D in the preceding diagram. If the remote codec **Send** LQ reading is 01 then the outgoing audio from the remote codec is being blocked by a firewall at either point A or B in the preceding diagram.

TROUBLESHOOTING TCP PORT BLOCKING

Error messages on the codec screen can help to diagnose TCP port blocking.

1. **"Connection Refused"** usually means that the firewall is configured correctly but the codec is not using the expected port. For example, the firewall is set up to forward via port 9002 but codec is 'listening' to port 10,000. “Connection Refused” is not normally shown if the firewall is not configured correctly because a firewall will by design silently drop any forwarding requests to ports that it doesn’t have open (see next point). Note: "Connection Refused" will also be displayed if the Commander G3 or i-Mix G3 codec you are calling is already connected.
2. **“Connection Timeout”** can mean one of two things:
   - The firewall is not configured correctly and the attempted codec connection is being silently dropped, e.g. a remote codec is dialing to port 9002 but the studio firewall port forwarding is not configured.
   - The UDP port is not port forwarded correctly. Tieline codecs send test data during connection establishment to make sure that the audio path is configured correctly; if this process fails then it will also result in a “Connection Timeout”.

HOW DO I DETERMINE WHICH END IS BLOCKING DATA FLOW?

Tieline test codec firewalls have the default Tieline TCP and UDP ports open. You can dial into these test codecs (or other codecs you know are configured correctly) from your recently configured studio and remote codecs and use the LQ readings to diagnose whether your studio or remote codec firewall is blocking your data packets. If one codec connects ok and the other one doesn't, then you will know which end is likely to be causing the problem. As an example:

1. Dial from site 1 to a Tieline test codec.
2. Dial from site 2 to Tieline test codec.

If both of these connect successfully then the “outbound” TCP path for session data is OK, and the inbound UDP audio path is OK.

1. Dial to site 1 from a codec you know is configured correctly.
2. Dial to site 2 from a codec you know is configured correctly.

If either of these calls fail then TCP and/or UDP inbound data is being blocked on the failed connection (see "Troubleshooting TCP Port Blocking" previously).

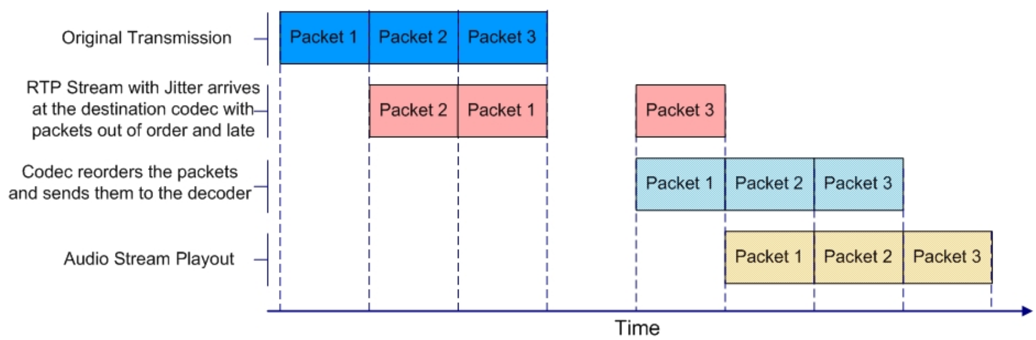
DEFAULT TIELINE CODEC PORT SETTINGS

Firewall Ports							
Commander G3 / i-Mix G3		Bridge-IT / Bridge-IT XTRA		Merlin and Genie Codec Families		ViA Codec	
TCP	UDP	TCP	UDP	TCP	UDP	TCP	UDP
IP1 Session Port: 9002	IP1 Audio Port: 9000	Session Port (Sess): 9002	Audio (Proto): 9000	Session Port: 9002	Audio Port Stream 1: 9000	Session Port: 9002	Audio Port Stream 1: 9000
IP2 Session Port: 9012	IP2 Audio Port: 9010	Web-GUI: 80	SIP Session: 5060	Alternative Session: 9012	Audio Port Stream 2: 9010	Alternative Session: 9012	Audio Port Stream 2: 9010
Toolbox Software: 5550	Toolbox Software: 5550	Alternative Session: 9012	SIP Audio: 5004	Web-GUI: 80	Audio Port Stream 3: 9020	Web-GUI: 80	SIP Session: 5060
	SIP Session: 5060	Alternative Web-GUI: 8080	Fuse-IP 8999	Alternative Web-GUI: 8080	Audio Port Stream 4: 9030	Alternative Web-GUI: 8080	SIP Audio: 5004-5054
	SIP Audio: 5004	TLS/SSL 443		TLS/SSL 443	Audio Port Stream 5: 9040	TLS/SSL 443	Fuse-IP 8999
					Audio Port Stream 6: 9050		
					SIP Session: 5060		
					SIP Audio: 5004-5054		
					Fuse-IP 8999		

WHAT IS CAUSING PACKET LOSS?

Before discussing packet loss it is useful to understand the concept of 'jitter' and the jitter buffer settings available in Tieline IP codecs.

Jitter, also known as latency or delay, is the amount of time it takes for a packet of data to get from one point to another. A jitter buffer is a temporary storage buffer used to capture incoming data packets. It is used in packet-based networks to ensure the continuity of audio streams by smoothing out packet arrival times during periods of network congestion. Data packets travel independently and arrival times can vary greatly depending on network congestion and the type of network used, i.e. LAN versus wireless networks. The concept of jitter buffering is displayed visually in the following image.



Tieline jitter-buffer technology can:

- Remove duplicate packets.
- Re-order packets if they arrive out-of-order.
- Repair the stream in the event of packet loss (error concealment).
- Manage delay dynamically based on current network congestion.
- Manage forward error correction (FEC).

With Tieline codecs you can configure either a fixed or automatic jitter buffer and the settings you use depend on the IP network over which you are connecting. Over LANs, WANs and wireless networks the automatic jitter buffer generally works well. It adapts automatically to prevailing IP network conditions to provide continuity of audio streaming and minimize delay. A fixed jitter buffer is preferable over satellite or high latency connections to ensure continuity of signals.

CAUTION: If a Tieline codec connects to a device that is using non-compliant RTP streams then the last fixed setting configured in the codec will be enabled (default is 500ms). Non-compliant devices include some other brands of codec, web streams and other devices.

TIELINE 'AUTO JITTER BUFFER' SETTINGS

The following automatic jitter buffer settings range from the most aggressive "Least Delay" setting, which endeavors to minimize delay as much as possible while adapting to prevailing network conditions, to the "Highest Quality" setting, which is the most conservative delay setting to minimize packet loss when packet latency is not as critical. Best compromise is the most popular setting used by most broadcasters as it achieves an excellent balance that minimizes latency for bidirectional communications, as well as minimizing packet loss over most wired and wireless IP networks.

Least Delay: This setting attempts to reduce the jitter buffer to the lowest possible point, while still trying to capture the majority of data packets and keep audio quality at a high level. This setting is the most aggressive in adapting to prevailing conditions, so the jitter buffer may vary more quickly than with the other settings. It is not recommended in situations where jitter variation is significant, or occurs in bursts. (E.g. cellular/multi-user wireless networks). It is best for stable and reliable links such as dedicated or lightly-loaded WAN/LANs.

Less Delay: This setting lies between "Best Compromise" and "Least Delay". It may assist in reducing latency over a connection without incurring packet loss.

Best Compromise: This default setting is the midpoint between the jitter buffer settings applicable for "Highest Quality" and "Least Delay." It is designed to provide the safest level of good audio quality without introducing too much latency. In most situations it will deliver very high quality and low delay to support live bidirectional communications over cellular and wireless networks.

Good Quality: This settings lies between "Best Compromise" and "Highest Quality." It may assist in achieving higher quality connections without incurring extreme delays in transmission or significant packet loss.

Highest Quality: This setting is the most conservative in terms of adapting to prevailing network conditions to reduce delay. The jitter-buffer will remain higher for a longer period after a jitter spike is detected – just in case there are more spikes to follow. This setting is best to use where audio quality is the most important factor and delay is not as critical. Unless delay is irrelevant, this setting is not recommended over peaky jitter networks (e.g. cellular networks) and is best used on more stable networks where fluctuating jitter bursts are not common.

JITTER DEPTH

The jitter **Depth** setting allows you to select predetermined minimum and maximum jitter settings within the auto jitter buffer's minimum and maximum jitter limitations. The default setting of **60 to 1000ms** is a good starting point for most networks. It may be necessary to increase the maximum auto jitter latency setting for networks experiencing higher packet latency, or the minimum depth depending on the reliability of the network.

PACKET LOSS STATISTICS

Tieline codecs display packet loss statistics as follows:

	Feature	Description
1	Lost Packets	Packets sent that failed to arrive
2	Empty (Jitter Buffer)	Indicates how often the jitter buffer ‘reservoir’ empties causing loss of audio
3	Late Packets	The number of packets that arrive late, i.e. after audio play out
4	FEC Packets	Indicates the number of forward error correction (FEC) packets that have been replaced if it is enabled in the codec
5	1 minute	Statistics listed for the last minute of network activity
6	10 minutes	Statistics for the last 10 minutes of network activity

Following is a packet arrival analysis table with solutions for any noticeable packet loss statistics displayed on the screen.

Packet Analysis	Displays	Possible Causes	Possible Solutions
Loss	Packets failed to arrive.	• LAN/WAN congestion • Unreliable ISPs • Unreliable networks • Unreliable IP hardware	• Renegotiate connection bit rate downwards • If link quality good, add or increase FEC as required • Assess ISPs QoS if very bad performance
Empty	Indicates how often the jitter buffer 'reservoir' empties causing loss of audio.	• High number of packets being lost or arriving late • Signal dropouts using cell-phone networks • Renegotiation causes the jitter buffer reservoir to empty	• Once could be an anomaly – assess lost & late packets • If many lost packets and network is unreliable – renegotiate bit rate and /or FEC down • If many late packets, increase jitter buffer maximum and minimum depth
Late	The number of packets that arrive late and after audio play out.	• Network congestion • Jitter Buffer depth is too low	• Auto-jitter buffer will adjust automatically • For manual jitter buffer settings, increase jitter buffer depth 50-100 ms & reassess (if only a few packets arrive late over time; audio repairs will be automatic and may not require buffer changes).
FECd	Indicates the number of FEC repaired packets if FEC active.	• Packets have been lost or corrupted over the network	• Assess audio quality & the number of FEC repairs – if many packets are being 'lost', perhaps reduce FEC &/or renegotiate bit rate down.

Important Notes:

- Increasing the minimum jitter buffer depth may address the issue of packets arriving late.
- If the jitter buffer, FEC or the connection bit rate is changed, we recommend assessing a minute of recent connection performance in preference to 10 minutes of historical connection performance. 10 minutes of data will include connection settings which may no longer be relevant. 'Packet arrival history' is cleared when you hang up a connection.