VCOM Network Bandwidth Requirements Guide

The network bandwidth requirements must be carefully analyzed to ensure proper bandwidth is available at any point where multiple clients will share the same physical connection point. The most obvious connection point where this is critical is at the server where bandwidth requirements will be the sum of the requirements of every possible client. The least obvious connection point where this is also important occurs when multiple remote clients in one physical location need to access the server in another physical location as the bandwidth requirements for the connection between these two points will be the sum of the requirements for all remote clients.

To determine the bandwidth requirements it is necessary first to determine the network bandwidth utilization per client connection, which is indicated below for the various audio sample rates that can be configured.

Audio Sample Rate	Data Rate (Kbps) [ATS=20ms*]	Data Rate (Kbps) [ATS=40ms*]	Data Rate (Kbps) [ATS=60ms*]	Data Rate (Kbps) [ATS=80ms*]	Data Rate (Kbps) [ATS=100ms*]
8 KHz	32	23.6	20.8	19.4	18.56
16 KHz	44.8	36.4	33.6	32.2	31.36
32 KHz	46.8	38.4	35.6	34.2	33.36

To determine server bandwidth requirements, first determine maximum potential bandwidth utilization by multiplying the number of clients (users and devices interfaced) by the Data Rate associated with appropriate Audio Sample Rate for the configured Audio Time Slice per packet. The product is the bandwidth required if every client were to receive audio simultaneously (maximum download bandwidth requirement) and also the bandwidth required if every client were to send audio simultaneously (maximum upload bandwidth requirement). In a typical system, the maximum download bandwidth requirement must be allocated for, as several system functions can require simultaneous audio transmission to all clients. The maximum upload bandwidth requirement however will realistically never be achieved as it is not feasible that all audio sources in a system would be active simultaneously since the result would be inaudible. As such the upload bandwidth to be allocated must be made based on the estimation of the number of simultaneous active audio sources noting that inactive audio sources will have no bandwidth requirements.

*ATS = Audio Time Slice per packet which controls how many 20ms audio frames are transmitted within a single UDP packet. As each UDP packet requires a fixed amount of overhead, the more frames sent at the same time, the less the UDP overhead which conserves network bandwidth. Conversely, the more audio frames sent per transmission, the greater the system latency and the potential audible consequence of a lost packet. The default is 20ms.

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