STRENGTHENING THE SIGNAL

# PTP 550 for VoIP Applications

## THE COMMUNICATIONS OF TODAY

From landline to cellular, communication technology continues to evolve. Voice over Internet Protocol (VoIP) is the voice and multimedia delivery method of the advancing digital age – transmitting communications from any computer or mobile device with a broadband connection. Increasingly popular VoIP applications including Skype, Google Talk, and WhatsApp keep hundreds of millions of residential and commercial users streaming voice and video to socialize, game, telecommute, and more.

VoIP works by converting sounds into digital signals, then sending them as data over the internet (connected via fiber cables or between towers connected wirelessly) to the recipient, where the digital signal is then reconverted back into sound. The technology is convenient for communication across any distance provided there's internet access, with few limitations to how many users can join a call or how many minutes calls last – but it has some drawbacks. Two of the most significant challenges for VoIP applications are call quality and security.

## THE WEAK POINTS

VoIP calls can struggle with "jitter" – a variety of glitchy symptoms including silent lag periods, choppy conversation, and crackling noises that impact sound clarity – caused by issues with the transmission of the packets of data that comprise the digitized sound input. The delays, interruptions, or changes to the delivery pace or order of packets can happen at any point along the IP packet path.

The security of VoIP calls is only as good as that of the applications that support them. The strength of encryption is as important for the protection of the conversations and data being sent via VoIP as it is for any other form of digital communication, such as email or online banking.

## THE IDEAL SOLUTION

The Cambium Networks PTP 550 backhaul solution redefines the potential of VoIP, with powerful capabilities specialized for stable, reliable high-speed connections for short and middle range applications – ideal to address the challenges of both quality and security.

Within a single IP67-rated enclosure, the PTP 550 is composed of two radios that can operate independently of one another – each on a separate frequency and channel size – with dynamic channel selection that continuously seeks and selects the channel that offers optimal available spectrum to ensure uninterrupted transmission. Each radio is also independently able to choose channel size.

The 100,000+ packet processing per second capacity of the PTP 550 meets the demand for low latency that streamlines VoIP calls. Automatic repeat request is another feature that safeguards call quality, enabling the PTP 550 to request an immediate re-transmit of an IP packet dropped between 550 links – rather than the call suffering any noticeable interruption. Each PTP 500 radio is equipped with AES128 encryption as a standard feature, offering VoIP calls end-to-end encryption for all packets once enabled, keep conversations private and data secure.

## PROVEN SUCCESS

To prove VoIP performance, PTP 550 is put under two test scenarios in which throughput and call quality are measured to ensure that the VoIP connection remains stable despite both high amounts of other types of data traffic and noise conditions.

### TEST 1

**5 VoIP calls of 30 seconds each are made back-to-back under the following conditions:** 88 dBm of Noise is injected while 160 Mbps of data is sent uplink on a 40 MHz channel size. Voice quality is measured using the ixLoad tool.

**FIGURE 1A:** Voice call made at 36 SNR (RSSI 52 dBm and Noise injected at 88 dBm) (Clean channel) with about 150 Mbps of IP traffic received in uplink.

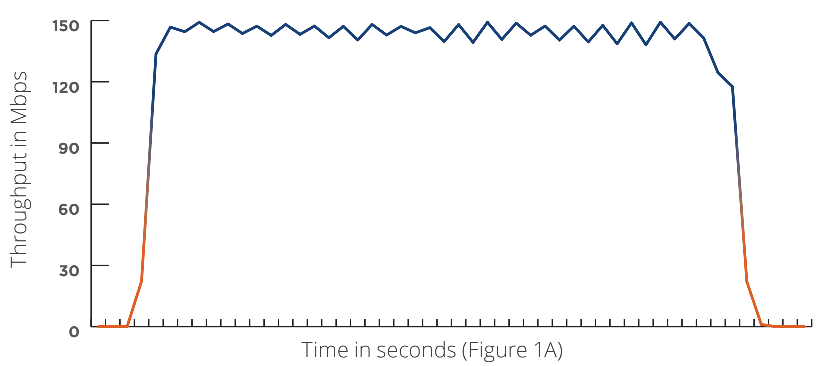
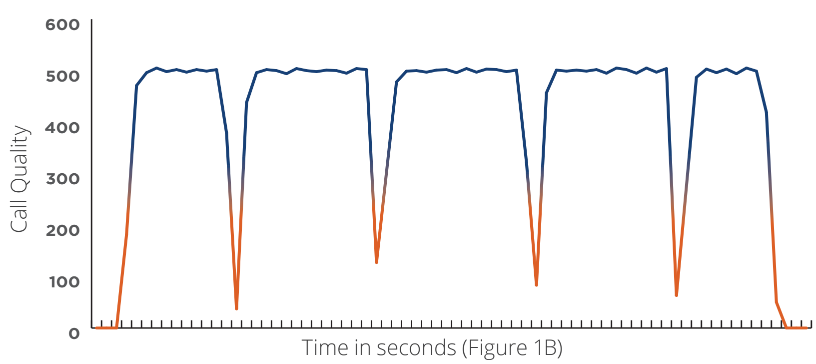



**FIGURE 1B:** Call quality is excellent (with 0 being unconnected, 1-200 low, 200-400 average, and 401-500 high call quality).



Results:

- Throughput remains relatively constant, fluctuating between 80-95% load capacity.
- Call quality is excellent – throughput dips only during the beginning and end of the call.
- Average Jitter latency is very low – no audio lags expected.

TEST 2

**Test 1 is repeated with 71 dBm of noise injected.**

**FIGURE 2A:** Voice call made at 19 SNR (RSSI 52 dBm and noise injected at 71 dBm) (Noisy channel) , with about 110 Mbps received in uplink

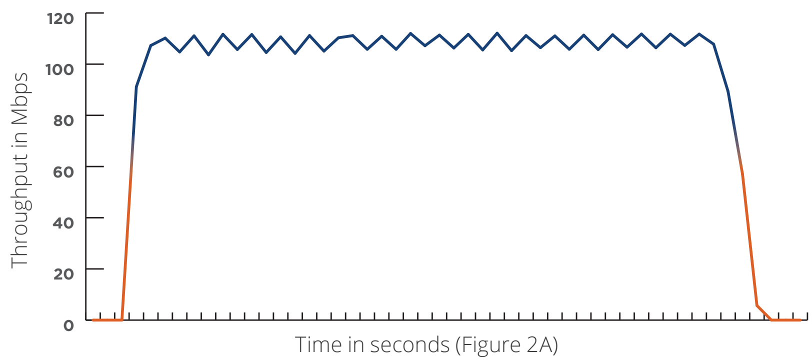
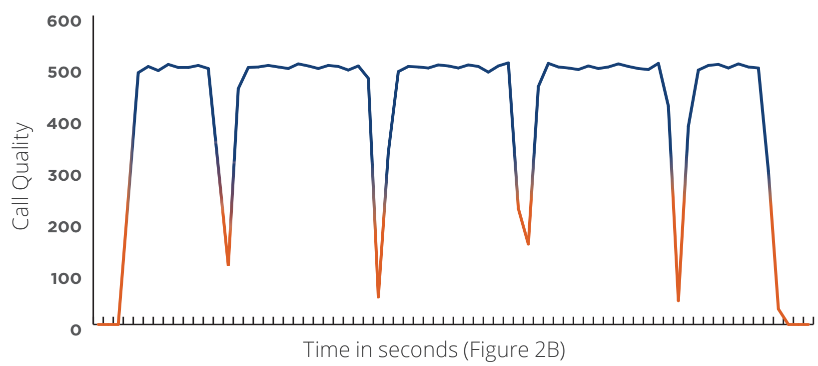



**FIGURE 2B:** Call quality test as in 1B repeated with 19 SNR.




Results:

- Throughput remains relatively constant, fluctuating between 80-95% load capacity.
- Call quality is good throughout the duration of each call, resulting in no perceptible issues.

## SUMMARY

From the two cases, it is shown that when interference is introduced, though the throughput level drops and the audio quality shows slight volatility, the overall call performance and user experience delivered by PTP 550 is successful despite the adverse interference conditions.