



AUTO ATTENDANT / RESEARCH & PRACTICAL GUIDANCE

VoIP Bandwidth per Call: Codecs, Jitter and Latency

Get a business phone number with a simple auto attendant menu from Auto Attendant. A caller picks who they need and the call rings that person's own mobile, so nobody on your team has to publish a personal number.

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The short answer

Plan for about **87 kbps in each direction for every simultaneous call**. That is Cisco's figure for a G.711 call (the classic uncompressed codec) over Ethernet with the usual 20 ms packets: 87.2 kbps per call ([Cisco, Modify Bandwidth Consumption Calculation for Voice Calls](#)). G.729 needs 31.2 kbps by the same table, but you rarely control which codec gets negotiated, so G.711 is the safe planning number.

For most small offices, bandwidth is not the problem: five simultaneous G.711 calls come to roughly 0.44 Mbps each way. What breaks calls is **delay, jitter and packet loss**, usually because something else is filling the upload side of the connection.

Figures are as of October 2026, from the documents linked inline. This is educational information, not legal or engineering advice.

Why a codec's "bit rate" isn't the bandwidth

G.711 is a 64 kbps codec, so why does a call use 87.2 kbps? Because the voice is chopped into small packets and every packet carries headers. Cisco's calculation ([Cisco](#)):

- **Voice payload:** with 20 ms of audio per packet, G.711 puts 160 bytes of voice in each packet.
- **Packets per second:** 20 ms per packet means 50 packets every second.
- **IP/UDP/RTP headers:** 40 bytes per packet (IP 20, UDP 8, RTP 12).
- **Ethernet headers:** 18 bytes per packet, including the 4-byte frame check.

So each packet is $160 + 40 + 18 = 218$ bytes, and $218 \text{ bytes} \times 8 \text{ bits} \times 50 \text{ packets per second} = 87,200$ bits per second. That is one stream of packets. A conversation has a stream each way, so your upload carries one and your download the other. Look at **upload** speed, often the smaller number on a plan.

The 20 ms packet is the standard default: RFC 3551 says the default packetization interval "SHOULD have a duration of 20 ms or one frame, whichever is longer," and lists 20 ms for PCMU and PCMA (the two G.711 variants), G.722 and G.729 ([RFC 3551, July 2003](#)).

Bigger packets save bandwidth but add delay: Cisco shows a G.729 call dropping from 24 kbps to 16 kbps (before Ethernet headers) when the payload doubles from 20 ms to 40 ms, while "the overall delay increases" (Cisco).

Don't count on silence suppression. Cisco says Voice Activity Detection "must not be taken into account" for bandwidth planning, "especially on links that carry fewer than 24 voice calls simultaneously" (Cisco).

G.711 vs G.729 vs Opus

Codec	Codec bit rate	Per call on Ethernet, 20 ms packets	Notes
G.711 (PCMU/PCMA)	64 kbps	87.2 kbps	Uncompressed PCM, standard since 1988
G.722	64 kbps	87.2 kbps	Wideband (16 kHz sampling)
G.726	32 kbps	55.2 kbps	Compressed
iLBC (20 ms mode)	15.2 kbps	38.4 kbps	Compressed
G.729	8 kbps	31.2 kbps	Compressed (CS-ACELP)
Opus (speech)	about 8-40 kbps	not in Cisco's table	Variable bit rate

Sources: Ethernet figures from Cisco; G.722 sampling rate from RFC 3551; Opus rates from RFC 7587.

G.711 is pulse code modulation (ITU-T G.711, 11/88, in force). Uncompressed, it is the heaviest common codec but also the most widely supported. The WebRTC audio standard requires G.711 alongside Opus, to ensure "voice interoperability capabilities between WebRTC endpoints and legacy phone systems that support G.711" (RFC 7874, May 2016).

G.729 codes speech at 8 kbit/s (ITU-T G.729, 06/12, in force). Its bit rate is one-eighth of G.711's, but because headers stay the same size, the call on Ethernet uses about a third (31.2 vs 87.2 kbps).

Opus is a newer, variable-rate codec that supports 6 to 510 kbit/s. With 20 ms frames, RFC 7587 gives speech "sweet spots" of 8-12 kbit/s for narrowband, 16-20 kbit/s for wideband and 28-40 kbit/s for fullband (RFC 7587, June 2015). RFC 7874 recommends offering Opus before G.711 when a device handles more than 8 kHz audio, but don't assume Opus carries your call end to end: a call to an ordinary phone number may switch codecs along the way. To estimate an Opus call on Ethernet, add Cisco's 58 bytes of headers per packet (23.2 kbps at 50 packets per second) to the codec rate; that is our arithmetic, not a published figure.

Takeaway: your provider, phone or app usually negotiates the codec. Size for G.711.

Sizing a small office

Simultaneous calls x 87.2 kbps = bandwidth needed in each direction (G.711 on Ethernet)

Calls at the same time	Upload needed	Download needed
1	87.2 kbps	87.2 kbps

Calls at the same time	Upload needed	Download needed
2	174.4 kbps	174.4 kbps
5	436 kbps	436 kbps
10	872 kbps	872 kbps
20	1.74 Mbps	1.74 Mbps

Calculated from Cisco's per-call figure. Encryption, VPNs and the link type change the overhead, so treat these as minimums.

How to count "simultaneous calls":

1. Count people who could be on the phone **at the same moment** at your busiest hour, not total staff.
2. Count only calls whose audio passes through the office connection: IP desk phones, softphones on office computers, mobiles on office Wi-Fi.
3. Calls answered on a mobile over the cellular network, or on a landline, don't use the office internet.
4. A hosted phone menu runs at the provider; your connection only carries the call once it rings a device on your network. Menu interactions are short, so they rarely add to the peak.
5. Leave headroom for backups, video meetings, file sync and guest Wi-Fi.

If you are still deciding how many lines you need, see [how many phone lines a small business needs](#).

Latency, jitter and packet loss

Latency (delay) is how long a packet takes to get from mouth to ear. Cisco, citing ITU-T G.114, says "the recommended one-way overall delay for voice is 150 ms. For a private network, 200 ms is a reasonable goal, and 250 ms must be the maximum" ([Cisco](#); [ITU-T G.114, 05/03](#)).

Jitter is variation in that delay: packets due every 20 ms arrive bunched, then late. Phones smooth this with a jitter buffer, but Cisco Meraki notes buffers "are not a long-term solution": large ones add delay, overflowing ones drop packets ([Cisco Meraki, What is Jitter?](#)). The standard measurement is defined in RFC 3550, the RTP specification: "interarrival jitter" is "the mean deviation (smoothed absolute value)" of the difference in packet spacing at the receiver versus the sender, reported in RTCP receiver reports ([RFC 3550, July 2003](#)). Speed-test "jitter" may not be calculated the same way.

Packet loss is audio that never arrives: clipped words, robotic sound, gaps.

Cisco's thresholds for voice (vendor guidance, not a formal standard):

Metric	Cisco threshold
One-way delay	150 ms or less
Jitter	under 30 ms
Packet loss	1% or less

Sources: [Cisco, Troubleshoot Network Related Audio Issues on Catalyst 9000 Switches](#); [Cisco Meraki](#), attributing the figures to Cisco's Enterprise QoS design guide.

What breaks calls in practice

Meraki: "The most common cause of jitter is network congestion at an egress point within a network" ([Cisco Meraki](#)). In a small office that is usually your router's upload side. Common culprits:

- **Large uploads.** Cloud backups, video uploads and big attachments can fill the upload for minutes; voice packets queue behind them.
- **Video meetings.** Video uses much more bandwidth than voice and competes for the same upload.
- **Wi-Fi.** A busy or weak signal adds delay and loss before packets reach the router. Cabled desk phones are more predictable.
- **VPNs and security appliances.** They add headers and processing; test calls with and without them.
- **Overloaded routers.** These can add jitter even when the line is fine.

QoS basics. Quality of Service tells your router to send voice first. RFC 4594 (informational) recommends a Telephony service class for IP telephony, marked with the Expedited Forwarding code point (EF, binary 101110), for traffic needing "very low delay, very low jitter, and very low packet loss" ([RFC 4594, August 2006](#)). For a small office:

1. Turn on QoS or "voice priority" in your router and prioritize the phones.
2. Set the router's upload limit slightly below your real upload speed so the queue builds where QoS can reorder it.
3. Don't expect markings to be honored across the public internet. Meraki suggests QoS "via a service provider as part of a managed service" if you need it end to end ([Cisco Meraki](#)).
4. Schedule backups and large syncs outside business hours.

How to test your line

- [] Run a speed test from a wired computer at your **busiest hour**. Note upload, download, latency and jitter.
- [] Compare upload speed with your sizing-table number plus everything else that uploads.
- [] Place a test call while a large file uploads. If audio breaks up only then, the upload is saturating.
- [] Repeat the test call on Wi-Fi and on a cable. A big difference points to the Wi-Fi.
- [] Check the call statistics screen on your IP phone or softphone, if it has one. Some show jitter and loss per call.
- [] Call a mobile and a landline, both directions. One-way audio problems point to one direction of the link.
- [] Keep the results with your outage plan; see the [business phone outage recovery worksheet](#).

Moving off copper lines? See [replacing copper phone lines for a business](#).

Mobile networks

When a call rings a staff mobile answered on the cellular network, that leg travels over the mobile carrier's network, not your office internet; its quality depends on signal and the carrier. Office bandwidth matters only when the mobile uses office Wi-Fi (a softphone app or Wi-Fi calling).

A 4G or 5G connection can serve as backup internet. A few calls need little bandwidth, but mobile-data latency and jitter vary with signal and load, so test first. For how a call moves between networks, see [how a business call travels](#).

FAQ

How much data does a VoIP call use per minute? By arithmetic from Cisco's G.711 Ethernet figure: 87.2 kbps x 60 seconds is about 654 KB per minute in each direction, or roughly 1.3 MB per minute for both directions together (about 78 MB per hour). G.729 at 31.2 kbps works out to about 234 KB per minute each way. Actual metered usage varies with the codec, encryption and link type.

Is my 100 Mbps plan enough? For voice alone, very likely, but check the upload figure, which on many plans is far below the advertised download, and test at your busiest hour.

Can I run business calls on 4G? Yes, the bandwidth is modest. Whether it sounds good depends on latency, jitter and loss where you are. Test before relying on it.

What jitter is acceptable? Cisco's guidance is under 30 ms, with one-way delay of 150 ms or less and loss of 1% or less.

Auto Attendant provides a US or Canadian business number with a menu of up to seven options, each ringing up to three phones in turn, for \$29 USD per line per month.

Sources

1. [Modify Bandwidth Consumption Calculation for Voice Calls](#), Cisco (updated December 11, 2023). Per-call bandwidth table, header sizes, formula, payload size trade-off, VAD guidance, G.114 delay figures.
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Source links

Links cited in this article, in order of first appearance. Full addresses are provided for readers using extracted text.

1. <https://www.cisco.com/c/en/us/support/docs/voice/voice-quality/7934-bwidth-consume.html>
2. <https://www.rfc-editor.org/rfc/rfc3551>
3. <https://www.rfc-editor.org/rfc/rfc7587>
4. <https://www.itu.int/rec/T-REC-G.711-198811-I/en>
5. <https://www.rfc-editor.org/rfc/rfc7874>
6. <https://www.itu.int/rec/T-REC-G.729-201206-I/en>
7. <https://www.itu.int/rec/T-REC-G.114-200305-I/en>
8. https://documentation.meraki.com/Wireless/Design_and_Configure/Architecture_and_Best_Practices/What_is_Jitter%3F
9. <https://www.rfc-editor.org/rfc/rfc3550>
10. <https://www.cisco.com/c/en/us/support/docs/switches/catalyst-9300-series-switches/220655-troubleshoot-network-related-audio-issue.html>
11. <https://www.rfc-editor.org/rfc/rfc4594>

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About Auto Attendant

Auto Attendant gives a small business one phone number and a recorded menu that sends each caller to the right person. The call rings the mobile that person already carries, using their normal dialler and their normal mobile plan. Pricing is a single flat monthly fee for the whole company rather than a charge for each user, and there is no app, desk phone or hardware to install.

One number, one menu, the phones you already own

A caller dials the business number, hears a short greeting and chooses an option. Each option can ring one mobile, ring several at once so whoever is free answers first, or try people one after another in a set order. There is no limit on how many people sit on the receiving end, because nobody is billed per seat. Setup is done by the owner in a few minutes: pick a number, write the greeting, and point each option at a mobile.

What it deliberately is not

An auto attendant routes a call; it does not answer one. Nobody at Auto Attendant speaks to callers, and no AI stands in for a receptionist. A business that wants its calls handled, messages taken and appointments booked wants an answering service, which is a different product at a different price. A business that needs call recording, queues, agent dashboards or CRM integration wants a contact centre platform. Auto Attendant is for the case where the call should simply reach the right pocket.

Keeping personal numbers off the internet

The business number is the only number a caller ever sees. Personal mobile numbers are never displayed and never given out, which is what lets an owner put a number on a website, an invoice, a van or a Google Business Profile without handing a personal line to everyone who finds it. An existing advertised number can be ported across and kept, and the old line keeps working while the port is in progress.

Work with Auto Attendant

Visit [Auto Attendant](#) to see [how it works](#), [how it compares](#) to a personal mobile, Google Voice and a full phone platform, and the current [pricing](#).

Telephone number portability, emergency calling and caller-ID rules differ by country and change over time. Articles published here describe general practice and cite primary regulator and carrier sources; they are not a guarantee of any timeline, and they are not legal or regulatory advice for a specific business.

Website: <https://autoattendant.io>

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