

PROFESSIONAL
HD AUDIO QUALITY
POWERFUL FEATURES
EASY ACTIVATION



Why Simple Conferencing?

Incredible Value

The bottom line is savings, and with Simple Conferencing's flat rate pricing that costs less than \$30 per month, you can save over 50% on your conference calling expenses. With the explosion in cell phone use and the increasing adoption of unlimited calling plans, there's no need to pay for toll-free charges for all your callers HD Audio Quality.

Real HD Audio Quality

With support for HD Audio and our sound enhancement technology, Simple Conferencing is setting a new quality standard in audio conferencing.

Convenience

There's no need to schedule or reserve calls anymore with Simple Conferencing – your bridge is ready when you are, 24x7.

Two Ways to Connect

SimpleSignal provides a number of convenient ways to connect into conference calls.

- 1. Toll Access** - You'll be given a direct dial number to access your conference bridge any time 24/7.
- 2. Extension Access** - SimpleSignal Customers may dial a 3 digit extension which will directly prompt them to start a conference call

Simple Conferencing Features

Simple Conferencing provides all the features you're looking for in a conferencing service. Default feature settings can be established for each conference bridge. Most features can be accessed or configured during live conference calls by using your telephone keypad.

- Reservationless Conference Start
- High Conference Capacity - up to 100
- Entry/Exit Notification
- Self-Muting
- Conference Termination
- Participant Count
- Private Roll Call
- Multiple Hosts
- Touchtone (DTMF) Suppression
- Security Lock
- Simple Keypad Commands

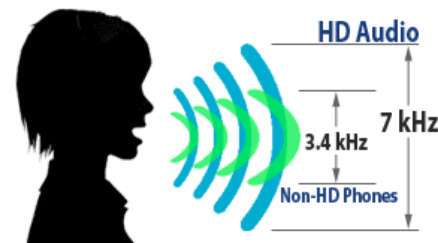


About HD Audio

Enabling Better Voice Quality

Did you know the basic speech technology in traditional phone networks has not changed in 75 years? But with new digital technology sound quality has been greatly enhanced, by more intelligently utilizing the abundant bandwidth available through today's broadband connections.

Simple Conferencing is at the epicenter of this evolution, delivering HD Audio quality by incorporating the ITU Standard wideband speech codec called G.722. Combined with Simple Conferencing's sound enhancement technology you can experience the highest quality in audio conferencing. Delivering a naturally-sounding conversation via audio conferencing requires sophisticated technology, normalizing sound levels, filtering background noises, and balancing audio streams. With HD Audio, the intelligibility of speech is greatly enhanced, and individual voices are easier to differentiate and understand, more closely simulating the experience of being in the same room around a conference table.

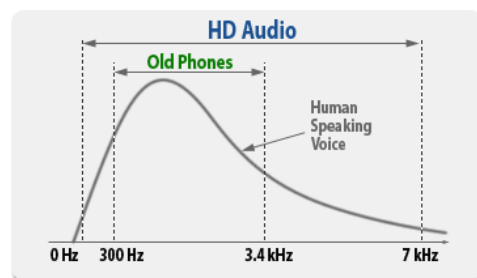


What are Codecs and How Do They Affect Sound Quality?

Sound quality is determined by the amount of data that is digitally sampled, transmitted and repackaged to reproduce the speaker's voice. Historically, telephone networks have used technology (called codecs) to minimize the amount of bandwidth required for transmission -- so they could carry more calls using the available capacity. Given the relative scarcity of wireless spectrum, mobile phone companies use more aggressive compression technology, which results in poorer sound quality. (To provide better quality would require either more spectrum, which may not be available, or smaller cell sites requiring many more towers.) That's why cell phones sound worse than landline phones, and why phone calls don't replicate your actual speaking voice.

A look at the numbers highlights the issue. There are two key factors:

- 1. Sample Rate.** This reflects how many times per second the sound is sampled. Audio CDs sample the sound 44,100 times per second (44.1kHz), reproducing a frequency range of 20 kHz, which is the limit of audible sounds to most humans.
- 2. Bit Rate.** This is the amount of bandwidth required. Sophisticated algorithms are used to minimize bandwidth, but the more aggressive techniques require more computations, leading to transmission delays (called latency). The effect of these techniques is to introduce delays between the time that the speaker talks and is heard on the other end. Fewer bits also means fewer "sound particles", which means greater deviation from the original sound.



During normal conversations, humans produce sounds from 80 Hz to about 8,000 Hz, with most normal speech occurring between 300 Hz and 3,000 Hz. (Singing or screaming can be outside this range, as you've no doubt noticed on phone calls.) Based on this, the traditional telephone networks were designed to transmit frequencies up to 3,400 Hz. The primary goal was to deliver sufficient quality to be understood, not to replicate speech quality.

There are three major codecs used by today's telephone networks:

- 1. G.711**, providing the best legacy telephone sound quality, but consuming the most bandwidth.
- 2. G.729**, used often for long distance transmission on landline networks, especially for overseas calls.
- 3. GSM-HR** ("HR" stands for "half-rate"), used by cellular companies.

The table below compares these codecs with the HD Audio codec, G.722:

Codec	Sample Rate	Min Frequency	Max Frequency	Bit Rate	Latency
GSM-HR	8 kHz	300 Hz	3.4 kHz	5.6 kbps	25 ms
G.729	8 kHz	300 Hz	3.4 kHz	8 kbps	15 ms
G.711	8 kHz	300 Hz	3.4 kHz	64 kbps	0 ms
G.722	16 kHz	150 Hz	7 kHz	64 kbps	4 ms

The bottom line is that G.722 provides a far superior audio sound with no noticeable latency, delivering a more natural conversation, with better clarity to discriminate between letters "S" and "F" or "P" and "T". This is especially true among female speakers, whose voices generate higher audio frequencies.

Using Simple Conferencing in HD Audio

To experience Simple Conferencing conference calls in HD Audio, you need three things:

- A Polycom HD Voice IP phone
- A broadband connection, since the G.722 codec is not enabled on legacy telephone networks.

Note: Any phone system will work with Simple Conferencing. Those conference attendees just won't experience the same HD Audio quality as the HD-enabled callers.