SIP Phones with High Sound Quality

Office phones continue to evolve. And as they do, the companies introducing SIP phones are growing. In addition to voice communication, SIP phones use IP protocol to exchange images and other data. This is quickly becoming the mainstream technology for office IP phones. However, while IP networks and other Internet devices are convenient, they don't guarantee the conversation quality. In other words, the critical function of sound quality is greatly affected by the performance of the phones themselves. If the quality is low, frequent problems such as constantly having to ask people to repeat themselves, or hearing something incorrectly, will prevent efficient communication. One way to solve these problems is to use G.722 audio coding. However, not all of today's SIP phones use G.722 or have designs that ensure high-quality sound. The level of sound quality is affected both by electrical sound reproduction factors, such as signal processing, and by acoustic factors, such as the ability of the speaker to carry the sound clearly to the listener's ear. Panasonic has accumulated massive amounts of data on both of these factors. Based on this data, we have literally filled our SIP phones with sound-enhancing technologies.

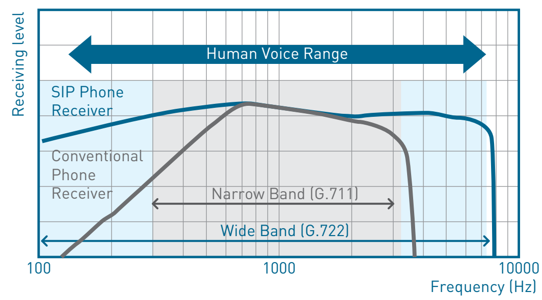
- •Using Digital Filters to Optimize Frequency Response
- •Incorporating Advanced Technologies in Speaker Units
- •Applying Echo-Canceling Technology
- •Resisting Network Faults

Panasonic supports comfortable offices and work styles with high-quality SIP phones. This White Paper presents a number of Panasonic technologies that were designed specifically to improve sound quality.

Using Digital Filters to Optimize Frequency Response

By tuning digital filters that support the G.722 codec, we have strengthened the characteristics of the low-frequency band below 300 Hz and the high-frequency band above 3.4 kHz. This has allowed us to provide an acoustically flat frequency response from 100 Hz to 7,000 Hz, for high-quality sound reproduction.

Receiving Frequency Response



Incorporating Advanced Technologies in Speaker Units

■ Speakers with Advanced Functions for Crisp, Clear Sound

•Two Types of Cone Paper

Low-frequency bandwidth output is increased by the use of two types of cone paper to make it easy to hear even low-pitched voices with excellent clarity.

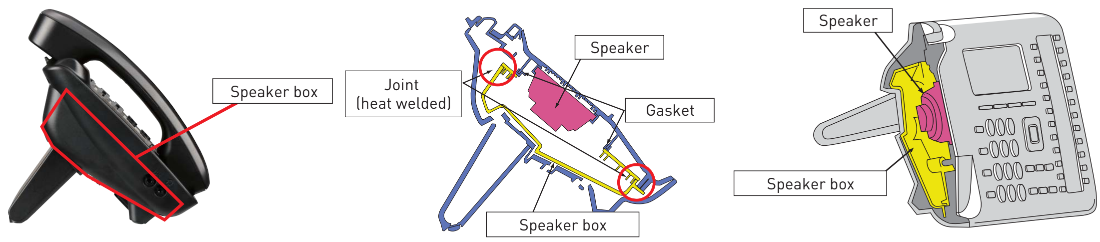
•Powerful Magnets

The use of a powerful magnet also raises the sound pressure across the entire bandwidth. This makes comparatively soft voices, which were previously difficult to hear, clear and easy to understand.

Unique speaker cone　Powerful magnet

Large-Volume Speaker Boxes

Large-volume speaker boxes enhance the reproduction of the low-frequency band (around 300 Hz). Using a speaker box also prevents the sound from leaking into the phone body, thus directing the sound toward the front with minimal loss. This achieves a high level of sound pressure. Echoes are reduced by preventing the sound of the speaker output signal inside the phone body from entering the phone mic. Because sound leaking inside the phone body lowers the sound volume that reaches the listener, a special gasket is used to prevent leaks from the surfaces where the speaker and phone body are in contact.



Optimization Simulations

Simply having a speaker box is not enough. Panasonic conducts research to determine the shape and size that will allow the speaker box to produce the kind of sound that is easiest to hear. Speaker boxes are created by varying the shape and size, and running simulations to find the optimal sound volume and quality.

Applying Echo-Canceling Technology

The KX-UT670 features a high-performance echo canceller driven by a powerful CPU to prevent echoes caused by the phone mic picking up sounds from the handset. This enables full duplex conversations. The KX-UT1xx Series also allows smooth, uninterrupted conversations.

Resisting Network Faults

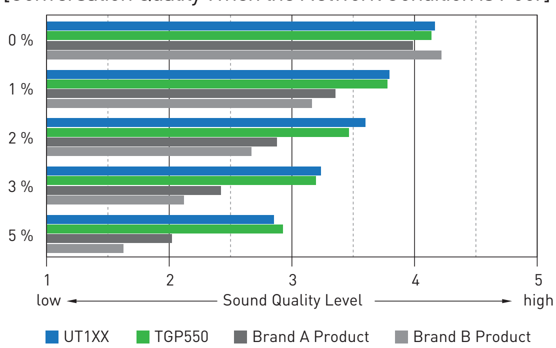
PLC(Packet Loss Concealment)

IP networks transmit audio data in units called packets. When a packet is "dropped," the listener hears it as a sound interruption or as static. The KX-UT Series uses a method called "packet loss compensation" to compensate for lost data and maintain the conversation quality. By using the optimal parameters to match the phone line quality, it is possible to produce sound that is closer to the original sound even when a packet loss occurs.

JB(Jitter Buffer)

The audio packets from the IP phone sending side are generally sent in regular intervals. However, when a delay of some kind happens while the packet is passing through the IP network, jitter occurs in the arrival time. In order to regularly reproduce the audio packets, the KX-UT Series uses something called a "jitter buffer" to absorb the jitter component. By using the optimal parameters to match the phone line quality, even when the line conditions are poor, it is possible to reduce delays and minimize the effects of sound interruptions and static.

[Conversation Quality When the Network Condition Is Poor]



The effect on sound when packet losses increase is lower for the KX-UT Series than for the competitor products.

Panasonic SIP Phones with High Sound Quality

Backed by a long track record of phone development, Panasonic's SIP phones support smooth and comfortable voice communication with the functions described above. The high sound quality of our SIP phones improves greatly upon the conventional weak points in phone conversation quality, to create a whole new form of business.

Panasonic SIP Phones also come in both corded and cordless types.
Users can select the type that best matches their particular office needs