Panasonic®

Operating Instructions

IP Conferencing Phone

Model No. **KX-NT700**

Thank you for purchasing this Panasonic product.
Please read this document carefully before using this product and save for future use.

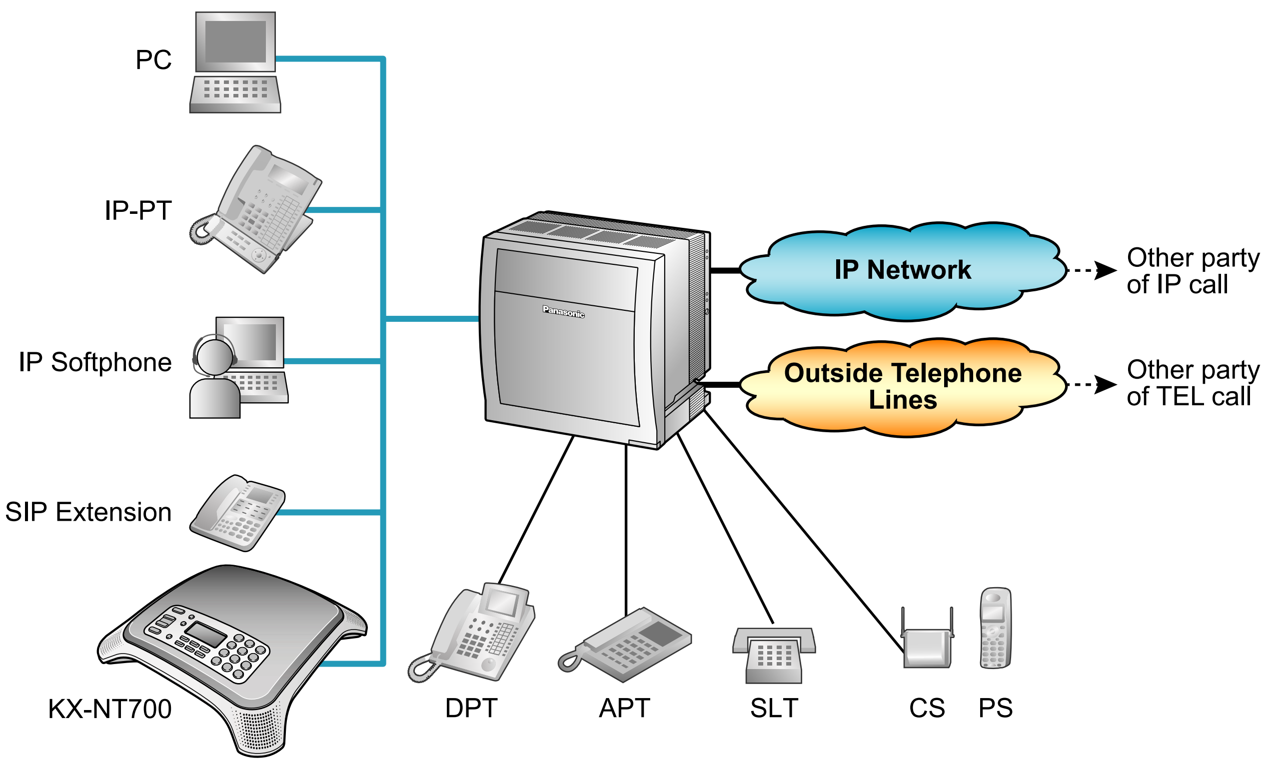
KX-NT700: Version 3.0 or later

Document Version 2011/06

Introduction

Connection to a Panasonic Pure IP-PBX (IP-PBX Mode)

The unit can be connected to a Panasonic KX-TDE or KX-NCP series PBX and used as a SIP extension. This allows you to make and receive calls using the outside lines and IP network connected to the PBX, call other extensions of the PBX by dialing their extension numbers, participate in conference calls with 4 or more other participants, etc.

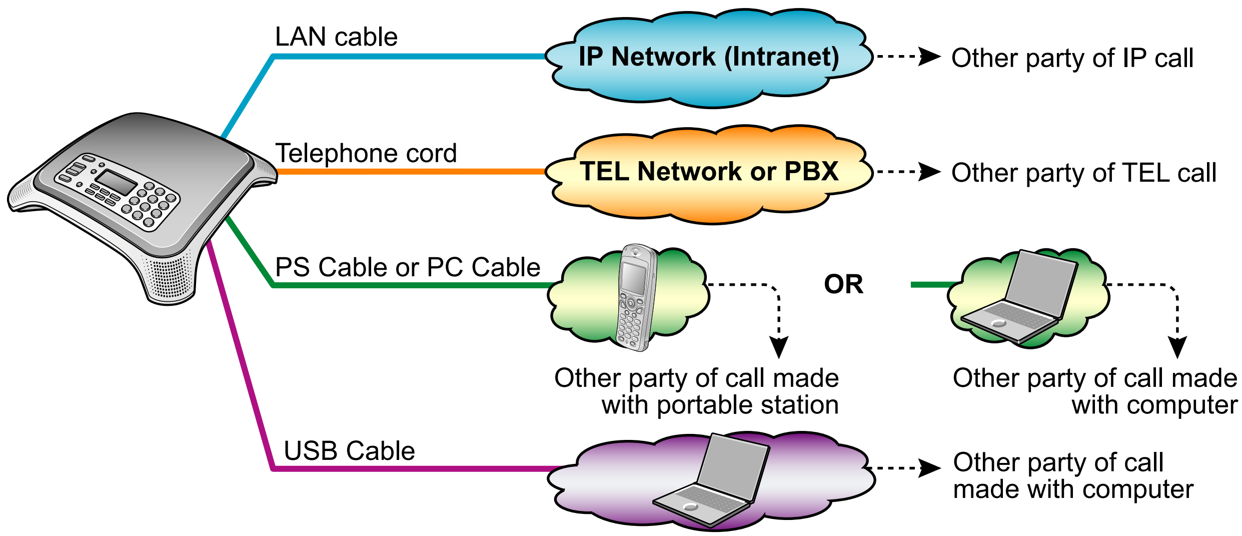


The following features are available when the unit is used as a SIP extension. Refer to the PBX documentation for details.

- Account Code Entry
- Automatic Route Selection (ARS)
- Conference (as a member only)
- DND Override
- Doorphone Call
- Extension Block
- Operator Call
- Personal Speed Dialing
- Redial
- S-CO Line Access
- System Speed Dialing
- TIE Line Call
- Trunk Group Access

Standard Connection Methods

Allow you to make and receive peer to peer IP calls, TEL calls, and PS or PC calls.



Connection to an IP Network[*1] (Intranet) and/or PBX

Allows you to make and receive calls over an IP network.
In this document, this connection method is referred to as the "IP line", and calls made using the IP line are referred to as "IP calls". There are 2 modes for making and receiving IP calls. When using IP-PBX mode, the unit can make and receive IP calls as a SIP extension of a Panasonic KX-TDE or KX-NCP series PBX (see page 2), or general PBX. When using peer to peer mode, the unit communicates directly with the other party's device.

[*1] If the IP network contains a firewall, the firewall must be configured appropriately to allow the unit to communicate over the network. See page 58 for information about the unit's VoIP communication settings.
If the IP network contains a router that supports NAT/NAPT features, it may not be possible for the unit to communicate over the network.
Consult your system administrator for details.

Connection to an Analog Telephone Network or PBX

Allows you to make and receive traditional phone calls.
In this document, this connection method is referred to as the "TEL line", and calls made using the TEL line are referred to as "TEL calls".

Connection to a Compatible Panasonic Portable Station

By connecting the unit to a compatible Panasonic Portable Station (PS) using the included PS Cable, you can use the microphones and speaker of the unit for calls made or received with the PS.
In this document, this connection method is referred to as the "PS line", and calls made using the PS line are referred to as "PS calls".

Connection to a Computer

By connecting the unit to a computer using the included PC Cable or the included USB Cable, you can use the microphones and speaker of the unit for calls made or received with the computer using your preferred IP phone software.
In this document, this connection method is referred to as the "PC line", and calls made using the PC line are referred to as "PC calls".

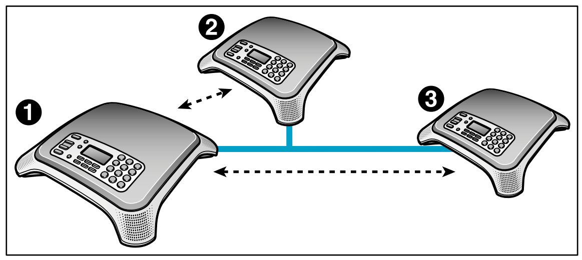
Other Features

Conference Calls

While on a call, you can make or receive an additional call, creating a 3-party conference call (see page 36 or page 38). Conference calls can be made using the following connection methods.

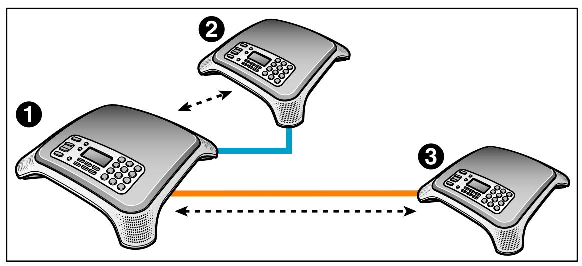
2 IP calls

While ❶ and ❷ are talking, ❶ calls or is called by ❸.



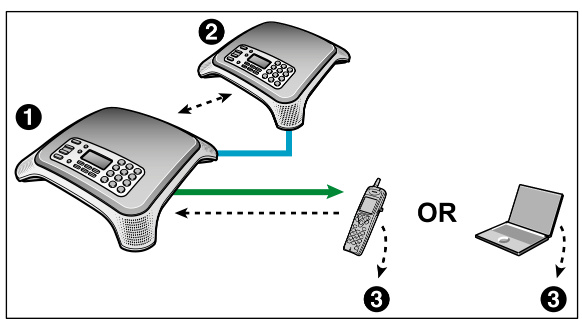
1 IP call and 1 TEL call

While ❶ and ❷ are talking, ❶ calls or is called by ❸.



1 IP call and 1 PS call or 1 PC call

While ❶ and ❷ are talking, ❶ uses a PS or computer to call ❸.



High-quality Audio

The unit provides unparalleled audio quality and features, including:

– G.722 speech codec support
– full-duplex communication
– speech speed conversion (see page 39)
– mic noise reduction (see page 39)
– External Wired MIC connection (see page 32)

SD Memory Card Recording

Phone calls and voice memos can be recorded to, and played back from, a compatible SD memory card (see page 42).

PoE (Power over Ethernet) Ready

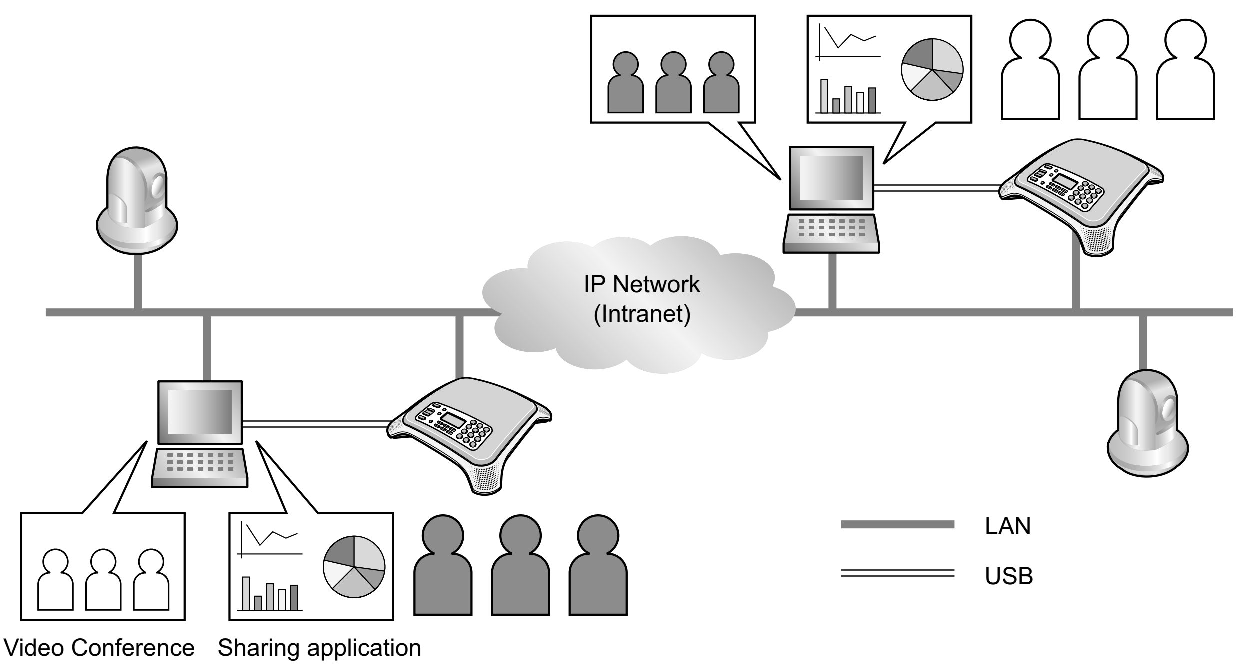
The unit is compliant with PoE (IEEE 802.3af) standards, and contains a power receiving device that enables it to receive power from the same Ethernet cable used for IP network connection. This allows you to use the unit in locations where there is no AC outlet nearby, saving you the cost of installing a new AC outlet. PoE connection requires a PoE-compliant hub or similar device. The included AC adaptor can be used instead of a PoE connection if you want to connect the unit to a standard AC outlet.

Conferencing Phone Manager Software Features

The unit can be used in conjunction with Conferencing Phone Manager. This software can be found on the included CD-ROM, and allows you to operate and program the unit using a computer (see the Operating Instructions for Conferencing Phone Manager for more details).

Video Conference/Sharing application

By using Conferencing Phone Manager, you can create a video conference and share applications with the other party.



Other Information

Included Documentation

Quick Reference Guide
Briefly describes how to connect the unit and introduces commonly used features.

Operating Instructions (this document)
Describes how to connect, use, program, and maintain the unit.

Operating Instructions for Conferencing Phone Manager
Describes how to operate Conferencing Phone Manager, which is computer software that can be used in conjunction with the unit.

Note

- Certain products and features described in this document may not be available in your area. Consult a certified Panasonic dealer for more information.

PBX Connection

- If the unit is connected to a PBX, refer to the PBX documentation for information about making calls, receiving calls, and other features.
- Do not connect the unit to an analog telephone line to which other telephones are connected.

Trademarks

- SD Logo is a trademark of SD-3C, LLC.
- Windows Media® is either a registered trademark or trademark of Microsoft Corporation in the United States and/or other countries.
- QuickTime® is a trademark of Apple Inc., registered in the U.S. and other countries.
- All other trademarks identified herein are the property of their respective owners.

MD5 Copyright Notice

This software uses the Source Code of RSA Data Security, Inc. described in the RFC1321 (MD5 Message-Digest Algorithm).

NetBSD kernel Copyright Notice

This product uses a part of NetBSD kernel.
The use of a part of NetBSD kernel is based on the typical BSD style license below.

Firmware Notice

- The unit's firmware is protected by copyright laws and international treaty provisions, and all other applicable laws. It cannot be reverse engineered, decompiled or disassembled.

For Future Reference

Record the information in the space below for future reference.

Note

- The serial number of this product may be found on the label affixed to the bottom of the unit. You should note the serial number of this unit in the space provided and retain this manual as a permanent record of your purchase to aid in identification in the event of theft.

MODEL NO.	____________________
SERIAL NO.	____________________
DATE OF PURCHASE	____________________
NAME OF DEALER	____________________
DEALER'S ADDRESS	____________________
	____________________
	____________________
	____________________
DEALER'S TEL. NO.	____________________

For Your Safety

To reduce the risk of injury, loss of life, electric shock, fire, malfunction, and damage to equipment or property, always observe the following safety precautions.

Explanation of symbols

The following symbols are used to classify and describe the level of hazard and injury caused when the denotation is disregarded and improper use is performed.

⚠ **WARNING**

Denotes a potential hazard that could result in serious injury or death.

⚠ **CAUTION**

Denotes a hazard that could result in minor injury or damage to the unit or other equipment.

The following symbols are used to classify and describe the type of instructions to be observed.

⃠

This symbol is used to alert users to a specific operating procedure that must not be performed.

❗

This symbol is used to alert users to a specific operating procedure that must be followed in order to operate the unit safely.

⚠ WARNING

General Safety

⃠ Do not disassemble this unit. Only qualified personnel should service this unit. Disassembling the unit may expose you to dangerous voltages or other risks. Incorrect reassembly can cause electric shock.

⃠ Do not insert foreign objects into the unit.

⃠ Do not connect or disconnect the AC plug with wet hands.

❗ Disconnect the unit from the AC outlet, disconnect the LAN cable, and contact the dealer if:
- The AC adaptor cord or AC plug becomes damaged or frayed.
- The unit is exposed to rain, water, or any other liquid.
- The unit is dropped or damaged.
- Internal components are exposed due to damage.
- The unit does not operate properly.
- Performance deteriorates.

❗ Disconnect the unit from the AC outlet and disconnect the LAN cable if the unit emits smoke, an abnormal smell, or makes unusual noise. These conditions can cause fire or electric shock. Confirm that smoke has stopped and contact an authorized service center.

❗ Clean the AC plug periodically with a soft, dry cloth to remove dust and other debris.

Installation

⃠ Do not connect the unit to the AC outlet, AC extension cords, etc., in a way that exceeds the power rating of, or does not comply with the instructions provided with, the AC outlet, AC extension cords, etc.

⃠ Do not touch the unit, AC adaptor, AC adaptor cord, or telephone cord during a lightning storm.

⃠ Do not install telephone jacks in wet locations unless the jack is specifically designed for wet locations.

Do not touch uninsulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.

If using an AC adaptor, use only the included AC adaptor (PQLV206).

The AC adaptor should be connected to a vertically oriented or floor-mounted AC outlet. Do not connect the AC adaptor to a ceiling-mounted AC outlet, as the weight of the adaptor may cause it to become disconnected.

Only connect the unit to the type of electric power specified on the label affixed to the unit. Confirm the type of electric power supplied to the installation site if necessary.

Use caution when installing or modifying telephone lines.

Placement

Do not expose the unit to contact with liquids (rain, water, moisture, oil, etc.) or excessive smoke or dust. Do not subject the unit to excessive shock.

Do not allow anything to rest on the AC adaptor cord or LAN cable. Do not locate this unit where the AC adaptor cord or LAN cable may be stepped on or tripped on.

Place this unit on a flat surface. Serious damage and/or injury may result if the unit falls.

Allow 10 cm (3 15/16 in) clearance around the unit for proper ventilation.

⚠ CAUTION

Do not place heavy objects on top of this unit.

When the unit receives power from the AC adaptor, the AC adaptor is the main disconnect device. Ensure that the AC outlet is installed near the unit and is easily accessible, so that the unit can be disconnected from the AC outlet if necessary.

Disconnect the AC adaptor cord and all cables from the unit before cleaning. Clean the unit with a soft, dry cloth. Do not use liquid, aerosol cleaners, abrasive powders, or chemical agents to clean the unit.

The SD memory card poses a choking hazard. Keep the SD memory card out of reach of children.

When left unused for a long period of time, disconnect the unit from the AC outlet. When the unit receives power from a PoE power supply, disconnect the LAN cable.

Notice

- Read and follow all instructions, warnings, cautions, etc. including those marked on the unit.
- Before connecting the unit, confirm that the unit supports the intended operating environment.
- If the unit does not operate properly, disconnect the AC adaptor cord and LAN cable, then connect again.
- The unit may not operate in the event of a power failure. Ensure that a separate telephone, not dependent on local power, is available for use in case of emergency.
- Do not move the unit while it is in use.
- To prevent malfunction, deformity, overheating, rust, and discoloration, do not install or place equipment in the following types of locations:
  - Locations exposed to direct sunlight.
  - Locations where the temperature is less than 0 °C (32 °F) or greater than 40 °C (104 °F).
  - Locations where there is high humidity.
  - Locations where air ventilation is poor.
  - Locations that may be exposed to sulphurous gas, such as near hot springs.
  - Near devices that emit heat, such as heaters.
  - Near devices that emit electromagnetic noise, such as radios or televisions.
  - Near devices that emit high-frequency noise, such as sewing machines or welders.
- Do not place credit cards, ATM cards, or other magnetic cards near the unit. The magnets in the unit's speaker and microphones may damage magnetic cards.
- If an error message is shown on the unit's display, consult the network administrator.
- Satisfactory operation, interoperability, and compatibility cannot be guaranteed with all

equipment connected to the unit, nor with all services provided by telecommunications providers over networks connected to the unit.

For Best Performance

- Use the unit in a quiet room. Ambient noise of less than 50 dBA is recommended.
- Use the unit in a room with minimal echoing. Do not place the unit near walls, windows, partitions, etc.
- During the first 30 seconds of a TEL call, the unit adjusts itself for optimal sound quality. Speak in turns with the other party at the beginning of a conversation. (The time required varies depending on the condition of the telephone line and the audio characteristics of the room.) During this time, sound may cut out or fade in and out. This is normal.
- Do not obstruct the unit during calls. Keep your hands, as well as common objects such as folders, cups, and coffee pots away from the unit during calls.

Data Security

We recommend observing the security precautions described in this section, in order to prevent the following:
- loss, disclosure, falsification, or theft of user information
- unauthorized use of the unit
- interference or suspension of use caused by an unauthorized party

We cannot be responsible for damages resulting from the misuse of this product.

Note

- This product can be used to store and log user information. User information is defined as the following:
  - phonebook entry names, phone numbers, IP addresses, SIP extension numbers and SIP URIs
  - call history (redial list)
  - recordings stored on the SD memory card

Preventing Data Loss

- Use a computer to make periodic backups of recordings stored on the SD memory card.
- Keep a separate record of all information stored in the phonebook.

Preventing Data Disclosure

- Do not leave the unit or SD memory card in a location where it can be accessed or removed without authorization.
- Store backups in a secure location.
- Do not store sensitive personal information in the unit.
- In the following situations, make a record of information stored in the phonebook, initialize the unit (see page 65), and remove the SD memory card from the unit.
  - Before disposing of the unit
  - Before handing the unit over to a third party
  - Before having the unit serviced
- Make sure the unit is serviced by only a certified technician.

Preventing Data Disclosure Over the Network

- To ensure the security of private conversations, only connect the unit to a secure network.
- To prevent unauthorized access, only connect the unit to a network that is properly managed.
- Make sure all computers connected to the unit employ up-to-date security measures.

Additional Information

FCC Requirements

1. **Notification to the Telephone Company**
   This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA. On the bottom of this equipment is a label that contains, among other information, a product identifier in the format US: ACJTE04BKX-NT700. If requested, this number must be provided to the telephone company.
   If required, provide the telephone company with the following technical information:
   – Telephone numbers to which the system will be connected
   – Make: Panasonic
   – Model: KX-NT700
   – Certification No.: found on the bottom of the unit
   – Ringer Equivalence No.: 0.4B
   – Facility Interface Code: 02LS2
   – Service Order Code: 9.0F
   – Required Network Interface Jack: RJ11
2. **Ringer Equivalence Number (REN)**
   The REN is used to determine the number of devices that may be connected to a telephone line. Excessive RENs on a telephone line may result in the devices not ringing in response to an incoming call. In most but not all areas, the sum of RENs should not exceed five (5.0). To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company. The REN for this product is part of the product identifier that has the format US: ACJTE04BKX-NT700. The digits represented by 04 are the REN without a decimal point (e.g., 04 is a REN of 0.4). For earlier products, the REN is separately shown on the label.
3. **Incidence of Harm to the Telephone Lines**
   If this equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.
4. **Changes in Telephone Company Communications Facilities, Equipment, Operations and Procedures**
   The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.
5. **Trouble with this Equipment**
   If trouble is experienced with this equipment, for repair or warranty information, please see the attached warranty, which includes the shipping address of the Panasonic Service and Technology Company BTS Center. If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.
6. **Connection to Party Line**
   Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.
7. **Combined Use with Alarm Equipment**
   If your home has specially wired alarm equipment connected to the telephone line, ensure the installation of this equipment does not disable your alarm equipment. If you have questions about what will disable alarm equipment, consult your telephone company or a qualified installer.
8. **Automatic Dialing Features**
   When programming emergency numbers and/or making test calls to emergency numbers:
   i. Remain on the line and briefly explain to the dispatcher the reason for the call before hanging up.
   ii. Perform such activities in the off-peak hours, such as early morning hours or late evenings.

Important Safety Instructions

When using this unit, basic safety precautions, including those below, should always be followed to reduce the risk of fire, electric shock and injury to persons.

1. Do not use the unit near water, for example, near a bathtub, wash bowl, kitchen sink, or laundry tub, in a wet basement, or near a swimming pool.
2. Avoid using a wired telephone during an electrical storm. There is a remote risk of electric shock from lightning.
3. Do not use the telephone in the vicinity of a gas leak to report the leak.

SAVE THESE INSTRUCTIONS

Required Telephone Cord

CAUTION

- To reduce the risk of fire, use only No. 26 AWG or larger telephone line cord.

Perchlorate Information

Notice

- This product contains a CR Coin Cell Lithium Battery which contains Perchlorate Material—special handling may apply.
  See
  www.dtsc.ca.gov/hazardouswaste/perchlorate

Interference

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

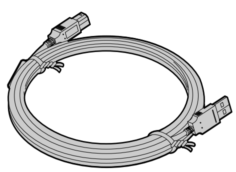
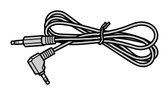
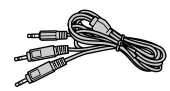
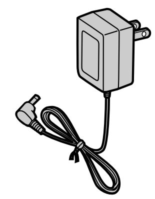
– Reorient or relocate the receiving antenna.
– Increase the separation between the equipment and receiver.
– Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
– Consult the dealer or an experienced radio/TV technician for help.

Table of Contents

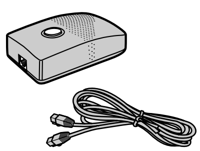
Accessory Information

Included Accessories

To order replacement accessories, call 1-800-332-5368.

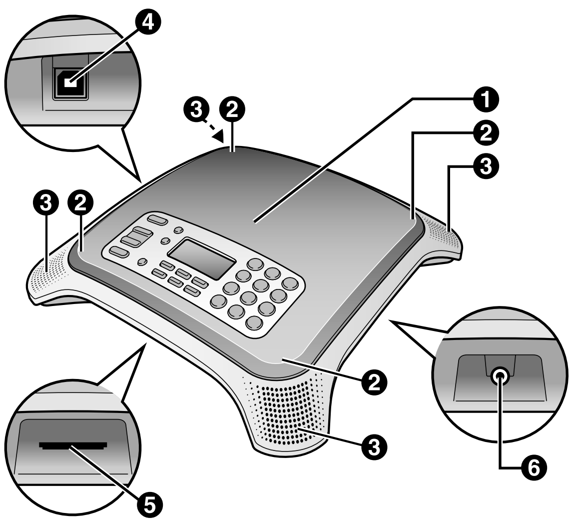
USB Cable (1) About 1.8 m (6 ft.)	**PS Cable (1)** About 1 m (3 ft. 3 in.)	**PC Cable (1)** About 1.8 m (6 ft.)	**AC Adaptor (1)**
			
Order no. PSWE2NT700N	Order no. PSJA1123Z	Order no. PSJA1122Z	Order no. PQLV206Y

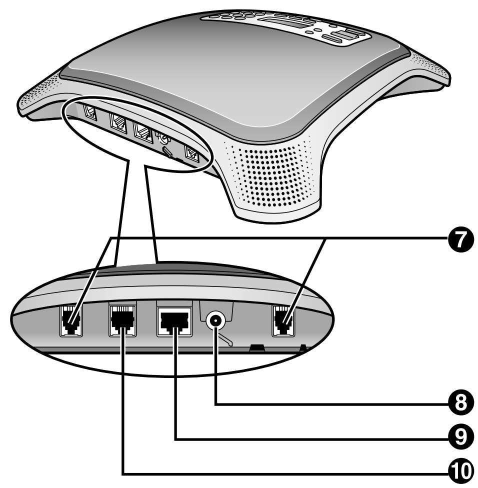
Optional Accessories

External Wired MIC Cord: About 3 m (10 ft.)

KX-NT701

Unit Overview

Main View





❶ **Speaker**

❷ **Indicators (4 locations)**
Indicate the status of the unit.
Off: The unit is in standby mode (i.e., not in use).
Blue, flashing: A call is being received.
Blue, lit: The unit is on a call.
Red, lit: The microphones are muted, or an error has occurred.

❸ **Built-in Microphones (4 locations)**
See page 32.

❹ **USB Port (USB)**
Used to connect the unit to a computer and use the included Conferencing Phone Manager software. Also used to connect the unit to a computer to use the microphones and speaker of the unit for your IP phone software (see page 45).

❺ **SD Memory Card Slot**
Allows you to insert a compatible SD memory card and record conversations. See page 30 for more information about SD memory cards.

❻ **AUDIO IN/OUT Jack**
Allows you to connect a compatible Panasonic Portable Station (PS; see page 44) or a computer (see page 46).

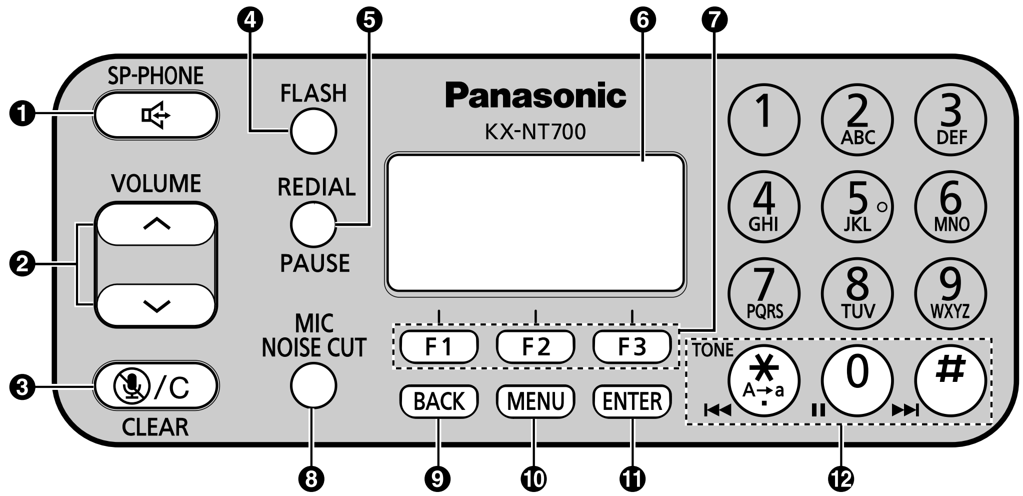
❼ **External Wired MIC Jacks (EXT MIC1, EXT MIC2)**
Allow you to connect an External Wired MIC to the unit (see page 32). 2 mics can be connected.

❽ **DC Input (DC IN)**
Used to connect the unit to an AC outlet using the included AC adaptor.

❾ **LAN Port (LAN)**
Used to connect the unit to an IP network. May also be used to supply power to the unit using PoE (Power over Ethernet) when the unit is connected to a PoE-compatible switching hub or power supply (see page 25).

❿ **Telephone Jack (LINE)**
Used to connect the unit to a telephone network or PBX.

Front Panel



❶ Button (Speakerphone Button)
Used to make, answer, and end calls.

❷ **Navigator/Volume Buttons ([^] and [v])**
Used to scroll through the items displayed on the display, such as phonebook entries, programmable settings, etc. Also used to adjust the speaker volume during calls (see page 35) and the ringer volume (see page 38).

❸ **/C Button (Mute/Clear Button)**
Used to erase characters or numbers while storing a phonebook entry or making a call. Also used to mute the unit's microphones during a call (see page 39).

❹ **FLASH Button**
Used to operate optional telephone company services, such as call waiting, or PBX features, such as extension transfers (see page 39).

❺ **REDIAL/PAUSE Button**
Used to call a previously called party again (see page 35) or to enter a dialing pause (see page 35).

❻ **Display**
See page 22.

❼ **Function Buttons ([F1], [F2], and [F3])**
Used to select the functions that correspond to the icons shown on the bottom of the display (see page 22).

❽ **MIC NOISE CUT Button**
Used to reduce noise in the audio signal sent to the other party during a call (see page 39).

❾ **BACK Button**
Used to return to the previous screen.

❿ **MENU Button**
Used to enter the programming menu or to return the unit to standby mode.

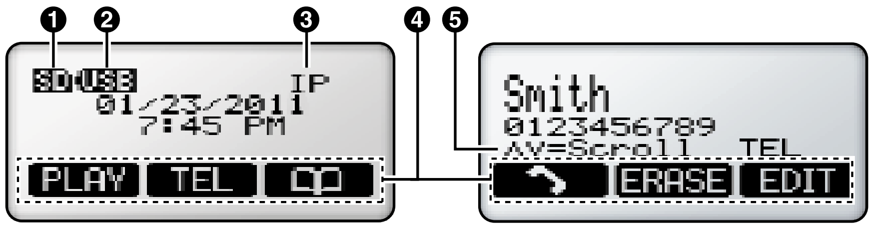
⓫ **ENTER Button**
Used to save or confirm information shown on the display.

⓬ **Playback Control Buttons**
Used to control playback when playing back recordings (see page 42).

Understanding the Display

The display helps you operate and program the unit by displaying a variety of messages and icons.

Standby Mode **Phonebook**



❶ **SD Icon**
Indicates that a compatible SD memory card has been inserted in the unit (see page 30).

❷ **USB Icon**
Indicates that a computer is connected to the USB port (see page 45).

❸ **Line Icon (IP, TEL, PS, PC, USB-AUDIO)**
Indicates which line will be used when a call is made.

❹ **Function Button Icons**
Indicates the functions currently available when the function buttons are pressed. The icons displayed vary on the current state of the unit (e.g., the icons displayed when on a call are different from the icons displayed when storing an entry in the phonebook).

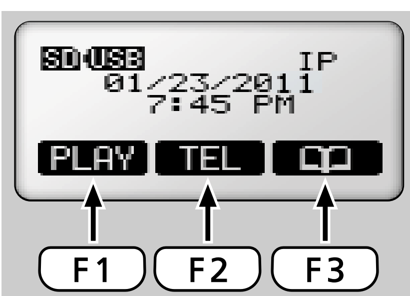
❺ **Scroll Indicator**
Indicates that **[∧]** or **[∨]** can be pressed to display the previous or next item.

Recording Icons

: Indicates a recorded conversation.
: Indicates a recorded voice memo.

Function Buttons and Function Button Icons

By pressing a function button (**[F1]**, **[F2]**, and **[F3]**) you can select the function displayed directly above it.



In this document, function buttons are referred to by their corresponding icons.
In the example shown here,
"Press PLAY ",
"Press TEL ", or
"Press "
would indicate pressing **[F1]**, **[F2]**, and **[F3]**, respectively.

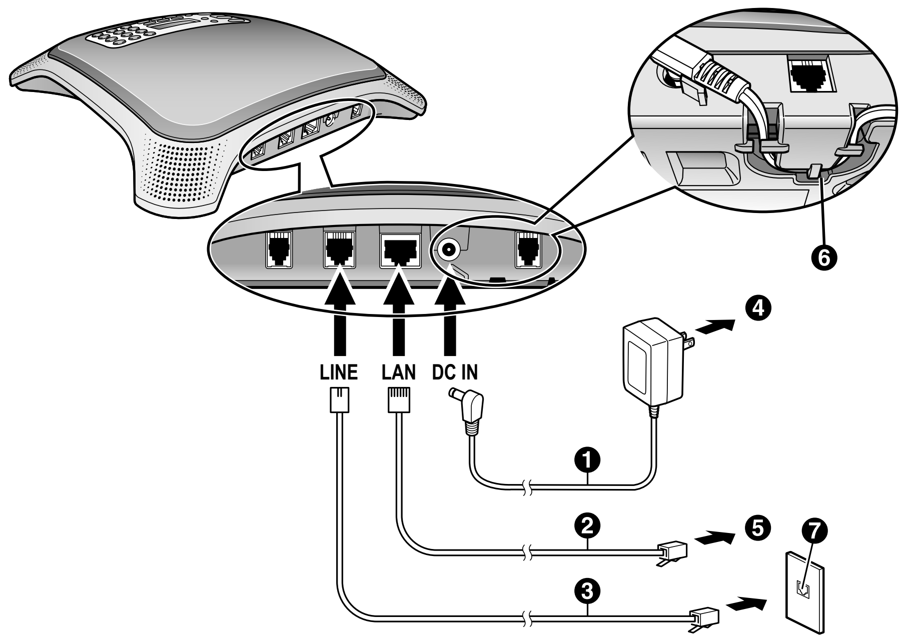
Function Button Icons

Operation	Icon	Description
Line Selection	TEL	Used to select the TEL line. Only displayed when the "**Line Selection**" setting is set to "**IP + TEL**" (see page 26).
	IP	Used to select the IP line.
	PS	Used to select the PS line. Only displayed when the "**Line Selection**" setting is set to "**IP + PS**" (see page 44).
	PC	Used to select the PC line. Only displayed when the "**Line Selection**" setting is set to "**IP + PC**" (see page 46).
	LINE	Used to change the "**Line Selection**" setting (see page 33).
Starting and Ending Calls	ANSWER	Used to answer an incoming call (see page 38).
	REJECT	Used to reject an incoming call (see page 38).
		Used to make a call (see page 34).
	END	Used to end the current call.
	CONF	Used to establish a conference call (see page 36 and page 38).
	SPEED	Used to slow down the other party's speech while on a call (see page 39).
	URI	Used when you want to enter a SIP URI. Only displayed when making a call on an IP line when the "**Operation Mode**" setting is set to "**IP-PBX**" (see page 34).
	EXT.	Used when you want to enter a SIP extension number. Only displayed when making a call on an IP line when the "**Operation Mode**" setting is set to "**IP-PBX**" (see page 34).
Phonebook		Used to open the phonebook (see page 40).
	ADD	Used to add an entry to the phonebook (see page 40).
	EDIT	Used to edit a phonebook entry (see page 41).
	CHAR	Used to switch between alphabet entry mode and extended entry mode (see page 40).
	<	Used to move the cursor to the left.
	>	Used to move the cursor to the right.
	ERASE	Used to erase an entry in the phonebook (see page 41).

Operation	Icon	Description
Recording and Playback	REC	Used to start recording to the SD memory card (see page 42).
	PLAY	Used to play the selected recording (see page 42).
	STOP	Used to stop recording (see page 42).
	◀◀	Used to rewind the current recording (see page 43).
	▶▶	Used to fast forward the current recording (see page 43).
	ERASE	Used to erase a recording (see page 43).
Other	BACK	Used to return to the previous screen.
	SELECT	Used to select the displayed item.
	SAVE	Used to save any changes made while programming the unit.
	YES	Used to accept the displayed item or proceed with the current operation.
	NO	Used to decline the displayed item or cancel the current operation.
	FORMAT	Used to format an SD memory card (see page 31).

Basic Connections

This section explains all connections needed to make and receive IP line and TEL line calls.
- To connect a Portable Station (PS) to the unit, see page 44.
- To connect a computer to the unit, see page 45 or page 46.



❶ AC Adaptor Cord
❷ LAN Cable
❸ 2-conductor Telephone Cord
❹ To AC Outlet
❺ To Switching Hub
❻ Groove
❼ Telephone Jack or PBX (RJ11C)

1. Connect the unit to the desired IP network and/or telephone line.
   - To connect to an IP network, connect a category 5 LAN cable to the **LAN** port and to a switching hub.
   - To connect to a telephone line, connect a telephone cord to the **LINE** jack and to a modular telephone jack.
2. Connect the AC adaptor cord of the included AC adaptor to the unit's DC input.
   - To use PoE (Power over Ethernet), connect the LAN cable to a PoE-compatible (IEEE802.3af) switching hub or power supply. The included AC adaptor does not need to be connected when using PoE.
   - If using an AC adaptor, use only the included AC adaptor (PQLV206).
   - Pass the AC adaptor cord through the groove on the bottom of the unit.
3. Connect the AC adaptor to the AC outlet.

Note

- The AC adaptor must remain connected at all times (unless the unit is powered by PoE). It is normal for the adaptor to feel warm during use.

Setting the Date & Time

Set the unit's date and time setting before using the unit. The date and time are shown on the display in standby mode, and are displayed when playing conversations that were recorded to an SD memory card.

1. Press **[MENU]**.
2. Select "**Basic Settings**", then press SELECT .
3. Select "**Date & Time**", then press SELECT .
4. Using the keypad, enter 2 digits each for the year, month, day of the month, hour (24-hour format), and minute.
   Example: To enter "Jan. 23, 2011, 7:45 PM", press **[1101231945]**.
   - If you make a mistake, press < to move the cursor as needed, then enter the correct number.
5. Press SAVE .
6. Press **[MENU]**.

Note

- You can select 12-hour or 24-hour time display (see page 52).
- The date format varies by the selection made for the "**Language**" setting. See page 51 for details.

TEL Line Settings

Selecting the Available Lines

To use the TEL line, the "**Line Selection**" setting must be set to "**IP + TEL**". (This is the default setting.)

1. Press **[MENU]**.
2. Press LINE .
3. Select "**IP + TEL**".
   - When "**Line Selection**" is set to "**IP + PC**" or "**IP + PS**", TEL calls cannot be made or received.
4. Press SAVE .
5. Press **[MENU]**.

Setting the Dial Mode

Set the dial mode to "**Pulse**" if the TEL line does not support tone dialing.

1. Press **[MENU]**.
2. Select "**TEL Settings**", then press SELECT .
3. Select "**Dial Mode**", then press SELECT .
4. Select "**Pulse**" or "**Tone**".
5. Press SAVE .
6. Press **[MENU]**.

IP Network Settings

To properly connect the unit to an IP network, the following settings must be set to match the settings of the IP network. Consult your system administrator for the appropriate settings.

– IP address mode: Automatic (DHCP) or manual (static) IP address assignment (default: static)
– IP address (when static connection mode is selected; default: 192.168.0.2)
– Subnet mask (when static connection mode is selected; default: 255.255.255.0)
– Default gateway (when static connection mode is selected; default: 0.0.0.0)

Note

- IP addresses can be entered using the keypad. **[0]**–**[9]** are used to enter numbers and **[∗]** is used to enter a period. For example, to enter "192.168.0.1", press **[192∗168∗0∗1]**.

Automatic Assignment (DHCP)

1. Press **[MENU]**.
2. Select "`IP Network Settings`", then press SELECT .
3. Select "`IP Address Mode`", then press SELECT .
4. Select "`DHCP`", then press SAVE .
5. Press **[MENU]**.

Note

- If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).
- To confirm the IP address, press **[MENU]**→"`Show IP Address`"→ SELECT .

Manual Assignment (Static)

1. Press **[MENU]**.
2. Select "`IP Network Settings`", then press SELECT .
3. Select "`IP Address Mode`", then press SELECT .
4. Select "`Static`", then press SAVE .
5. Select "`IP Address`", then press SELECT .
6. Enter the IP address to be assigned to the unit, then press SAVE .
7. Select "`Subnet Mask`", then press SELECT .
8. Enter the subnet mask, then press SAVE .
9. Select "`Default Gateway`", then press SELECT .
10. Enter the IP address of the default gateway, then press SAVE .
11. Press **[MENU]**.

Note

- If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).
- To confirm the IP address, press **[MENU]**→**[Show IP Address]**→ SELECT .

SIP Settings

To use the unit as a SIP extension of the PBX, the unit must be registered as a SIP extension through PBX programming, the "**Operation Mode**" setting must be set to "**IP-PBX**" (see page 29), and the following settings must be set to match the settings of the PBX. Consult your system administrator for the appropriate settings. Refer to the PBX documentation to register the unit as a SIP extension.

1. Press **[MENU]**.
2. Select "**Protocol Settings**", then press SELECT.
3. Select "**SIP Ext. No.**", then press SELECT.
4. Enter the unit's extension number (max. 32 characters), then press SAVE.
5. Select "**SIP Authentication ID**", then press SELECT.
   - If this field is left empty, the value set for "**SIP Ext. No.**" will be used as the SIP authentication ID.
6. Enter the SIP Authentication ID (max. 32 characters), then press SAVE.
7. Select "**SIP Password**", then press SELECT.
8. Press EDIT.
9. Enter the password (max. 32 characters), then press **[ENTER]**.
10. Select "**SIP User Domain Name**", then press SELECT.
11. Enter the IP address or domain name (max. 64 characters) of the PBX, then press SAVE.
12. Select "**SIP Proxy Server IP Address**", then press SELECT.
13. Enter the IP address of the PBX, then press SAVE.
14. Select "**SIP Proxy Server Port Number**", then press SELECT.
15. Enter the SIP port number of the PBX or SIP proxy server, then press SAVE.
16. Select "**SIP Registrar IP Address**", then press SELECT.
17. Enter the IP address of the PBX, then press SAVE.
18. Select "**SIP Registrar Port Number**", then press SELECT.
19. Enter the SIP port number of the PBX or SIP registrar server, then press SAVE.
20. Press **[MENU]**.

Note

- When the unit is connected to a device other than a Panasonic PBX, "**Panasonic PBX Compatibility**" in "**Protocol Settings**" must be set to "**Off**" (default: **On**).
- You can switch between numeric and alphabet entry modes by pressing CHAR, and change between uppercase and lowercase character entry by pressing **[*]**. See page 64 for information on entering characters.
- If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).
- To confirm the SIP extension number, press **[MENU]**→"**SIP Ext. No.**"→SELECT.

Correcting a Mistake

To correct a mistake, press < or > to move the cursor to the desired position, then follow one of the procedures below.

– To add a character or number, press the appropriate dial key.

– To erase the selected character or number, press **[/C]**.

To erase all characters and numbers, press and hold **[/C]**.

Operation Mode

By selecting an operation mode, the unit can be operated as a SIP extension of the PBX, a peer to peer IP conferencing phone, or a computer's USB audio device. The available line selection for each operation mode is as follows:

Operation Mode	Available Line Selection
IP-PBX	IP + TEL
	IP + PS
	IP + PC
Peer to Peer	IP + TEL
	IP + PS
	IP + PC
USB Audio	–

1. Press **[MENU]**.
2. Select "**Operation Mode**", then press SELECT.
3. Select the desired setting.
   - "**IP-PBX**": The unit can make and receive IP calls as a SIP extension of the PBX. To make a call, the other party's SIP extension number or SIP URI is specified. (This is the default setting.)
   - "**Peer to Peer**": Peer to peer IP calls are possible. To make a call, the other party's IP address is specified.
   - "**USB Audio**": The unit operates as the USB audio device of a computer (see page 45).
4. Press SAVE.
5. Press **[MENU]**.

Note

- If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).
- When the operation mode is set to "**USB Audio**", IP, TEL, PS or PC calls cannot be made or received.

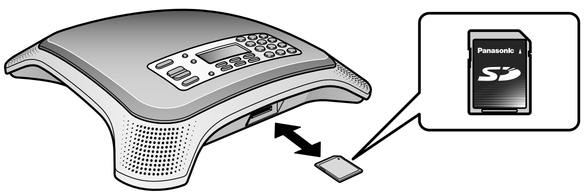
Using SD Memory Cards

Conversations can be recorded to the SD memory card. For information on recording conversations, see page 42.

Inserting and Removing Cards

Insert the SD memory card as shown, with the label side facing up. To remove the card, gently push the card to release it, then remove the card.

- When an SD memory card is inserted, SD is shown in the upper-left corner of the display.
- SD flashes while reading the data.



Important Information

To prevent data corruption or damage to the SD memory card, which may also affect the performance of the unit, keep the following in mind.

- Do not remove the SD memory card, LAN cable, or disconnect the unit from the AC outlet during playback, recording, formatting, reading, or while erasing data on the card.
- Do not move or bump the unit during playback, recording, formatting, reading, or while erasing data on the card.
- Do not touch the contacts on the bottom of the SD memory card.
- To prevent damage to the unit, do not insert any memory card other than a compatible SD memory card.

Compatible Cards

The unit supports the following SD memory cards.

– SD, miniSD, and microSD memory cards.
  - Use a miniSD or microSD adaptor when using miniSD or microSD memory cards, respectively, and always insert the miniSD or microSD memory card into the adaptor before inserting the adaptor into the unit.

– Cards with a capacity of 32 MB to 2 GB.

Note

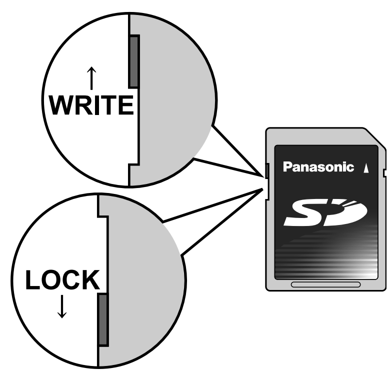
- The unit does not support SDHC, miniSDHC, and microSDHC memory cards.
- SD memory cards with a low minimum transfer rate may not be able to record conversations.

Approximate Recording Time

Capacity	Approx. Recording Time
2 GB	67 hours
1 GB	33 hours
512 MB	17 hours
256 MB	8 hours
128 MB	4 hours
64 MB	2 hours
32 MB	1 hour

Write-protection (LOCK)

SD memory cards can be locked to prevent formatting, erasing, and recording. To lock an SD memory card, slide the switch on the side of the card to the "LOCK" position.



Backing Up Data

Data stored on SD memory cards can become corrupted if the card is exposed to electromagnetic fields, static electricity, etc. We recommend using a computer to back up important data stored on SD memory cards.

Formatting SD Memory Cards

If **FORMAT** is displayed, the SD memory card must be formatted; press **FORMAT** to format the card.

Notice

- When an SD memory card is formatted, all information on the card is erased.
- Do not remove the SD memory card, LAN cable, or disconnect the unit from the AC outlet while formatting an SD memory card.
- Do not move or bump the unit while formatting an SD memory card.

Note

- The unit cannot format cards that are not already in FAT format. Use a computer to format non-FAT formatted cards.

Formatting With a Computer

When formatting cards with a computer, select the FAT (FAT16) format.

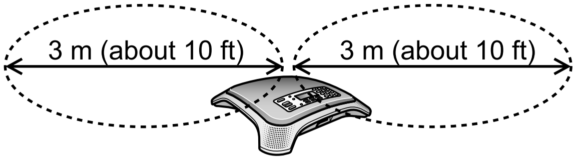
Notice

- When an SD memory card is formatted, all information on the card is erased.

Using the Microphones

Built-in Microphones

For best performance when using the built-in microphones, speak within about 3 m (about 10 ft.) of the unit.

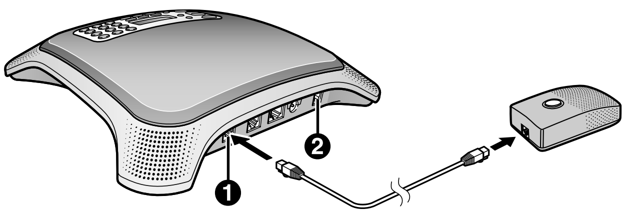


Note

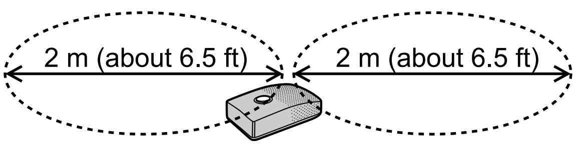
- The sensitivity of the built-in microphones may vary depending on room characteristics.

External Wired MIC

An optional KX-NT701 External Wired MIC can be connected to the unit using the **EXT MIC1** (❶) or **EXT MIC2** (❷) jacks. 2 mics can be connected.



For best performance when using an External Wired MIC:
– Do not move an External Wired MIC while on a call. (Feedback may occur.)
– Place each External Wired MIC at least 1 m (about 3 ft.) away from the unit.
– Speak within about 2 m (about 6.5 ft.) of the External Wired MIC.



Note

- The sensitivity of an External Wired MIC may vary depending on room characteristics.
- The built-in microphones continue to function when an External Wired MIC is connected.
- The indicator of an External Wired MIC indicates unit status the same as the built-in indicators (see page 20).

Line Selection

The unit can make calls using the following lines.

– IP line (see page 34)
– TEL line (see page 34)
– PS line (see page 44)
– PC line (see page 46)

When you make a call, the line icon in the upper-right corner of the display indicates the line that will be used, therefore, you should confirm the line icon each time you make a call.
You can change the selected line by pressing the center function button (**[F2]**). The line that will be selected is indicated by the function button icon.

Example:

1. The IP line is selected.

2. TEL is pressed. The TEL line is now selected.

Note

- If the unit is not connected to an IP network, IP is not displayed and the IP line cannot be selected.

Selecting the Available Lines

The IP line is always available when the unit is connected to an IP network; if it is not selected (i.e., if it is not shown in the upper-right corner of the display), you can select it by pressing IP .
Whether the TEL line, PS line, or PC line is available is determined by the "`Line Selection`" setting. For example, when it is set to "`IP + PS`", the IP and PS lines are available.
You can change the "`Line Selection`" setting using the following procedure.

1. Press **[MENU]**.
2. Press LINE .
3. Select the desired setting.
   - "`IP + TEL`": The IP and TEL lines are available.
   - "`IP + PS`": The IP and PS lines are available.
   - "`IP + PC`": The IP and PC lines are available.
4. Press SAVE .
5. Press **[MENU]**.

Note

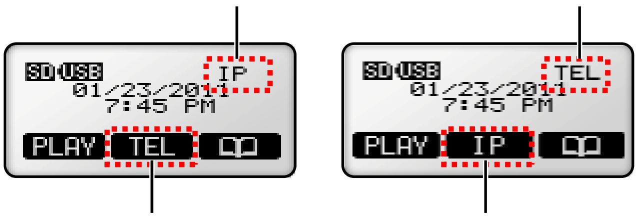
- When the operation mode is set to "`USB Audio`", the "`Line Selection`" setting is disabled.

Making Calls

Making a Call to a Phone Number or IP Address

1. Confirm that the desired line ("**IP**" or "**TEL**") is selected.
- You can change the selected line by pressing the center function button (**[F2]**).

The IP line is selected. The TEL line is selected.



[F2] **[F2]**

- If you cannot select the desired line, change the "**Line Selection**" setting (see page 33).

2. Press **[ ]**.

3. Enter the phone number or IP address.
- IP addresses can be entered using the keypad. **[0]**–**[9]** are used to enter numbers and **[*]** is used to enter a period. For example, to enter "192.168.0.1", press **[192*168*0*1]**. After entering the IP address, press **[#]** or **[ENTER]**.
- To temporarily use tone dialing when the line mode is set to pulse mode, press **[*]**.
- After a call starts, the approximate length of the call is shown on the display.

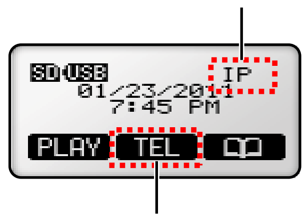
4. To end the call, press **[ ]**.

Note
- To make a call to a SIP extension number or SIP URI, see page 34.
- To make a call using the phonebook, see page 35.
- To make a call with a Portable Station (PS) or computer connected to the unit, see page 44 or page 45.
- During the first 30 seconds of a TEL call, the unit adjusts itself for optimal sound quality. Speak in turns with the other party at the beginning of a conversation. (The time required varies depending on the condition of the telephone line and the audio characteristics of the room.) During this time, sound may cut out or fade in and out. This is normal.
- If the unit is not connected to an IP network, IP is not displayed and the IP line cannot be selected.
- The call length shown on the display is an approximation and may differ from the actual length of the call. Call charges accumulate after the called party answers.

Making a Call to a SIP Extension Number or SIP URI

1. Confirm that the "**IP**" line is selected.
- You can change the selected line by pressing the center function button (**[F2]**).

The IP line is selected.



[F2]

- If you cannot select the desired line, change the "**Line Selection**" setting (see page 33).

2. Enter the SIP extension number or SIP URI.
- To enter a SIP URI:
  1. Press a dial key.
     The center function button icon will change to URI.
  2. Press URI to select SIP URI.
  3. "sip:" is displayed automatically. Enter the SIP URI after the colon.

     Example:

     sip:*User name**1@*2*Domain name**3

     (max. 32 characters) (max. 64 characters)

     *1 You can also enter a SIP extension number instead of a user name.

     *2 To enter @, press CHAR to switch the character entry mode to alphabet entry mode, then press **[#]** once.

     *3 You can also enter an IP address instead of a domain name.

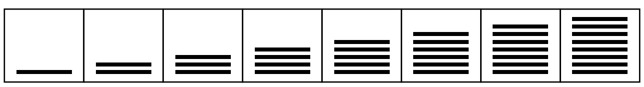
  If a domain name is not entered after the @, the value specified in "**SIP User Domain Name**" is used as the domain name when dialing (see page 56).

- You can switch between numeric and alphabet entry modes by pressing CHAR, and change between uppercase and lowercase character entry by pressing [*]. See page 62 for information on entering characters.
- A period can be entered by pressing **[1]**, when the character entry mode is alphabet entry mode.
- If you make a mistake, press **[/C]**, then enter the correct SIP extension number or SIP URI. To erase all numbers, press and hold **[/C]**.

3. Press **[ ]** or .
4. To end the call, press **[ ]**.

Adjusting the Speaker Volume

While on a call, press **[^]** or **[v]** repeatedly to adjust the speaker volume. There are 8 levels of volume.



- If the other party has difficulty hearing you, press **[v]** to decrease the speaker volume. Your voice heard by the other party will become louder.

Redialing

The last 10 calls made are stored in the redial list, in order of newest to oldest call.

1. Press **[REDIAL/PAUSE]**.
   - The last call made is displayed.
2. Press **[^]** or **[v]** repeatedly to scroll through the list.
   - To erase the displayed item, press ERASE.
   - To exit the redial list, press **[MENU]**.
3. When the desired entry is displayed, press **[ ]** or .

Note

- The line that was used to make each call in the redial list ("`IP`" or "`TEL`") is shown in the lower-right corner of the display. This line will be used when the call is redialed.
  Example:

- If the "**`Line Selection`**" setting is not set to "**`IP + TEL`**" (see page 33), TEL line calls cannot be redialed.
- If the dialed number contains too many characters, it cannot be redialed correctly. The maximum for each type of number is as follows:
  – TEL line: 128 digits
  – IP line (SIP extension number): 32 characters
  – IP line (SIP URI): 97 characters (excluding "sip:")

Entering Dialing Pauses

A pause is sometimes required when making calls on the TEL line using a PBX or a long distance service. For example, if you must dial "9" before dialing an outside phone number, you probably wait (pause) after dialing "9" until you hear a dial tone.
By pressing the **[REDIAL/PAUSE]** button when dialing, the unit will store the dialing pause along with the phone number in the redial list. If you make a call from the redial list later, the unit will dial the number wait for the pre-programmed number of seconds (default: 3 s; see page 54) for each dialing pause you entered.

Example:

1. Press **[9]** (to access an outside line of a PBX).
2. Press **[REDIAL/PAUSE]**.
   - Press **[REDIAL/PAUSE]** repeatedly to create longer pauses. An additional pause is inserted each time **[REDIAL/PAUSE]** is pressed.
3. Dial the phone number.
4. Press **[ ]** or .

Making a Call from the Phonebook

See page 40 to add entries to the phonebook.

1. Press .
2. Press **[^]** or **[v]** repeatedly to scroll through the phonebook entries.
   - Entries are displayed in the following order when **[v]** is pressed.
     Symbols→Numbers→Letters

- Press the dial key corresponding to the desired character, then press **[^]** or **[v]** to scroll if necessary.
- To exit the phonebook, press **[MENU]**.

3. When the desired entry is displayed, press **[ ]** or .

Note

- The line that was selected when the entry was stored in the phonebook ("**IP**" or "**TEL**") is shown in the lower-right corner of the display. This line will be used when the entry is called.

Example:

The TEL line will be used when you call this number.

- If the "**Line Selection**" setting is not set to "**IP + TEL**" (see page 33), TEL line numbers cannot be called.

Making Conference Calls

While on a call, you can make another call and establish a conference call (i.e., a 3-party call) including yourself and 2 other parties. You can establish a conference call using the following types of calls.

– 2 IP calls
– 1 IP call and 1 TEL call
– 1 IP call and 1 PS call
– 1 IP call and 1 PC call

To establish a conference call when you receive a call, see page 38.

Note

- When "**Panasonic PBX Compatibility**" in "**Protocol Settings**" is set to "**On**", you cannot make a conference call using 2 IP lines in "**IP-PBX**" mode (see page 58).

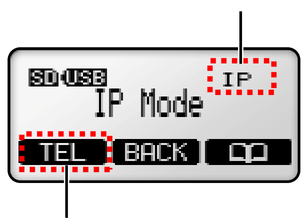
Conference calls using the IP and TEL lines

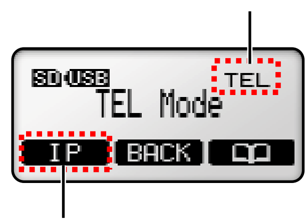
1. Press CONF to put the current call on hold.

2. Confirm that the desired line ("**IP**" or "**TEL**") is selected.
   - If the current call is an IP call, you can change the selected line by pressing the left function button (**[F1]**).

The IP line is selected.

The TEL line is selected.





[F1] **[F1]**

   - If you cannot select the desired line, press BACK, then change the "**Line Selection**" setting (see page 33). After you have changed the setting, repeat this procedure from step 1.

3. Call the party you want to add to the conversation.
   - You can end the second call and return to the original call by pressing BACK.
   - To call a party stored in the phonebook, see page 35.

4. After the called party answers, press CONF to begin the conference call.
   - Before beginning the conference call, press END to end the second call and return to the original call.

Note

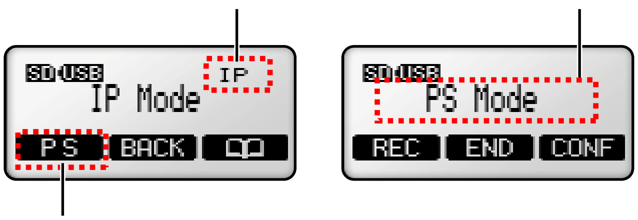
- We recommend setting the "**TEL Line Level Reduction**" setting to "**On**" when establishing conference calls that use the TEL line (see page 54).

Adding a PS or PC line call to an IP call

1. Confirm that the "**Line Selection**" setting is set to "**IP + PS**" or "**IP + PC**" as necessary (see page 33).
2. Press CONF to put the current call on hold.
3. Press the left function button (**[F1]**) to select the PS or PC line.

Example: Pressing PS to select the PS line.

The IP line is selected. The PS line is selected.



[F1]

4. Make a call using the PS or computer.
5. After the called party answers, press CONF to begin the conference call.
   - Before beginning the conference call, press END to end the second call and return to the original call.

Note

- When the operation mode is set to "**USB Audio**", conference calls cannot be made.

Ending a Conference Call

Press **[ ]** to disconnect both parties.

or

1. Press END.
   - Press BACK to continue the call.
2. Press **[^]** or **[v]** to select the party you would like to remove from the conference, then press SELECT.
   - The selected party is disconnected and you can continue to speak with the remaining party.
   - To disconnect both parties, select "**All**", then press SELECT.

Answering Calls

When a call is being received, the type of call being received is shown on the display.
Example: "**Incoming Call on IP Line**"

Notice

- When "**Line Selection**" (see page 33) is set to "**IP + PC**" or "**IP + PS**", TEL calls cannot be made or received.
- When the operation mode is set to "**USB Audio**", IP, TEL, PS or PC calls cannot be made or received.

1. Press [🔊] or ANSWER .
    - The unit does not support Caller ID; caller phone numbers are not displayed when calls are received.
    - You can refuse an incoming call by pressing REJECT .
    - After a call begins, the approximate length of the call is shown on the display.
2. To end the call, press [🔊].

Note

- During the first 30 seconds of a TEL call, the unit adjusts itself for optimal sound quality. Speak in turns with the other party at the beginning of a conversation. (The time required varies depending on the condition of the telephone line and the audio characteristics of the room.) During this time, sound may cut out or fade in and out. This is normal.

Adjusting the Ringer Volume

When the unit is in standby mode or is receiving a call, press [∧] or [∨] repeatedly to adjust the ringer volume. There are 4 levels of volume, including "**Off**".

Off	▁	▃	▇

Low High

Receiving a Second Call (Call Waiting)

While on a call, you can receive a second call, and then join the 2 calls and establish a conference call.
While on an IP call:
You can receive either a TEL or an IP call.
While on a TEL, PS, or PC call:
You can receive an IP call.

Note

- In order to use this feature, the "**Call Waiting**" setting (see page 53) must be set to "**Enable**" (this is the default setting).
- When a second call is received, a call waiting tone will be heard. See page 53 to adjust the call waiting tone volume.
- When "**Panasonic PBX Compatibility**" in "**Protocol Settings**" is set to "**On**", you cannot receive a second call on an IP line in "**IP-PBX**" mode (see page 58). You can only receive a second call on a TEL line.

Refusing a second call

Press REJECT . The second caller is disconnected and the current call continues.

Confirming the caller then creating a conference call

1. Press ANSWER .
    - The first call is put on hold, and you can talk to the second caller.
    - To end the second call, press END , then continue the first call.
2. Press CONF to establish a conference call.

Creating a conference call immediately

Press CONF .

Ending a conference call

Press [🔊] to disconnect both parties.
or

1. Press END .
    - Press BACK to continue the call.
2. Press [∧] or [∨] repeatedly to select the party you would like to remove from the conference, then press SELECT .
    - The selected party is disconnected and you can continue to speak with the remaining party.
    - To disconnect both parties, select "**All**", then press SELECT .

Useful Features Available During a Call

Mute

You can mute your voice during a conversation. While the mute is turned on, you will be able to hear the other party, but the other party will not be able to hear you.
To mute your voice, press **[/C]**. To return to the conversation, press **[/C]** again.

Note

- While the mute is turned on, "`Mute`" is displayed and the indicators light in red.
- All built-in microphones and each External Wired MIC are muted when the mute is turned on.

Flash

Pressing **[FLASH]** allows you to use optional telephone company services, such as call waiting, or PBX features, such as extension transfers.

Note

- To change the flash time, see page 54.
- This feature is not available for IP calls.

Speech Speed Conversion

You can adjust the speed of the other party's speech down during a call by pressing SPEED .
The following speech speed modes are available.

– **`Talking Mode`** (slight speed reduction)
  Recommended for calls in which you and the other party are equally participating in the conversation.
– **`Listening Mode`** (greater speed reduction)
  Recommended for calls in which the other party is speaking more, and you are listening.

You can select the desired mode by pressing SPEED during a call. Each time the button is pressed, the setting changes and is shown briefly on the display.

```
Conversion Mode
  Talking Mode
       ↓
Conversion Mode
 Listening Mode
       ↓
Conversion Mode
      Off
```

Note

- "`Slow`" is shown on the display while this feature is active.
- If the other party speaks for more than 5 seconds without stopping, this feature will stop functioning. Once the other party stops talking for about 1 second, this feature will function again.
- When this feature is turned off, the other party's speech may cut out briefly.

Mic Noise Reduction

You can press the **[MIC NOISE CUT]** button during a call to reduce the ambient noise that is picked up by the microphones and sent to the other party.
Each time the button is pressed, the setting changes and is shown briefly on the display.

```
Mic Noise Cut
    High
      ↓
Mic Noise Cut
     Low
      ↓
Mic Noise Cut
     Off
```

Note

- "`Noise Cut ON`" is shown on the display while this feature is active.
- The quality of the sound heard by the other party may decrease slightly while this feature is active, due to surrounding noise.

Adding Entries to the Phonebook

You can add up to 100 entries to the phonebook.
To make a call from the phonebook, see page 35.

1. Press [phonebook button].
   - The display shows the number of entries in the phonebook.
   - Entries cannot be added to the phonebook when the PS line or PC line is selected.
2. Press ADD.
3. Enter the name (max. 16 characters), then press **[ENTER]**.
   - See page 63 for information on entering characters.
   - To insert a space when there is no character under the cursor, press [>].
   - To insert a space after the last character entered, press [>] two times.
4. Press **[∧]** or **[∨]** to select the line ("**TEL**", "**IP(IP Address)**", "**IP(SIP Ext. No.)**", or "**IP(SIP URI)**") that will be used when you call the entry, then press SELECT.
   - When the operation mode (see page 29) is set to "**Peer to Peer**", select "**TEL**" or "**IP(IP Address)**".
     When the operation mode (see page 29) is set to "**IP-PBX**", select "**TEL**", "**IP(SIP Ext. No.)**" or "**IP(SIP URI)**".
5. Enter the phone number (max. 32 digits), IP address, SIP extension number (max. 32 characters) or SIP URI (max. 97 characters), then press **[ENTER]** or SAVE.
   - IP addresses can be entered using the keypad. **[0]**–**[9]** are used to enter numbers and **[∗]** is used to enter a period. For example, to enter "192.168.0.1", press **[192∗168∗0∗1]**.
   - If you selected "**IP(SIP URI)**", "sip:" is displayed automatically. Enter the SIP URI after the colon.

Example:

sip:*User name**1@*2*Domain name**3

(max. 32 characters) (max. 64 characters)

*1 You can also enter a SIP extension number instead of a user name.
*2 To enter @, press CHAR to switch the character entry mode to alphabet entry mode, then press **[#]** once.
*3 You can also enter an IP address instead of a domain name.

If a domain name is not entered after the @, the value specified in "**SIP User Domain Name**" is used as the domain name when dialing (see page 56).

   - A period can be entered by pressing **[1]**, when the character entry mode is alphabet entry mode.
   - To temporarily use tone dialing when the line mode is set to pulse mode, press **[∗]**.
   - If a pause is required when making a call on the TEL line (see page 35), press **[REDIAL/ PAUSE]** between digits as necessary.
   - To add an another entry, press ADD, then continue from step 3.
6. Press **[MENU]**.

Note

- If you do not press any buttons for 1 minute, the unit will return to standby mode.
- Only one type of destination (phone number, IP address, SIP extension number or SIP URI) can be stored in each phonebook entry.

Entering Characters

The dial keys are used to enter characters and numbers. Each dial key has multiple characters assigned to it. To enter a character, press the appropriate dial key, repeatedly if necessary. To enter another character that is assigned to the same dial key, first press [>] to move the cursor to the right.

Character Entry Modes

When adding entries to the phonebook, the following character entry modes are available. The current entry mode is shown in the upper-right corner of the display.
[ABC2]: Displayed when alphabet entry mode is selected.

[1234]: Displayed when numeric entry mode is selected.
[AÄÅ2]: Displayed when extended entry mode is selected.
Press CHAR to switch the character entry mode. Press **[*]** to change between uppercase and lowercase character entry.
See page 63 for a list of all available characters.

Correcting a Mistake

To correct a mistake, press < or > to move the cursor to the desired position, then follow one of the procedures below.

– To add a character or number, press the appropriate dial key.
– To erase the selected character or number, press **[🔇/C]**.

To erase all characters and numbers, press and hold **[🔇/C]**.

Editing Entries

1. Search for the desired phonebook entry (see page 35).
2. Press EDIT.
3. Edit the name if necessary, then press **[ENTER]**.
4. Press **[∧]** or **[∨]** to select the line ("**TEL**", "**IP(IP Address)**", "**IP(SIP Ext. No.)**", or "**IP(SIP URI)**") that will be used when you call the entry, then press SELECT.
5. Edit the phone number, IP address, SIP extension number or SIP URI if necessary, then press **[ENTER]** or SAVE.
6. Press **[MENU]**.

Note

- If you do not press any buttons for 1 minute, the unit will return to standby mode.

Erasing Entries

Erasing 1 Entry

1. Search for the desired phonebook entry (see page 35).
2. Press ERASE.
3. Press YES.
   - To cancel, press NO.
4. Press **[MENU]**.

Erasing All Entries

1. Press 📖.
2. Press ERASE.
3. Press YES.
4. Press **[MENU]**.

Note

- You can also erase entries using the "**Erase All Phonebook Data**" feature (see page 65).

Recording Features

Conversations and voice memos can be recorded to an SD memory card.

SD Memory Card Information

Confirm the following before recording to an SD memory card.

– A compatible card is inserted (see page 30)
  - When an SD memory card is inserted, SD is shown in the upper-left corner of the display.

– The card has been formatted using the correct format (see page 31).

– The card is not locked (see page 30).
  - If you insert a locked card, "**`Write Protected`**" is displayed.

Note

- No more than 100 recordings can be made, regardless of the SD memory card capacity.
- When "**`Memory Full`**" is displayed, recording is not possible until other recordings are erased. If the card becomes full while recording, recording will stop.
  See page 30 for information on approximate recording time.
- While recording, if the amount of recording time available is less than 6 minutes, "**`Remaining Time Less Than 6 Min.`**" is displayed briefly, and the display's backlight flashes until recording stops. When less than 1 minute is available, "**`Remaining Time Less Than 1 Min.`**" is briefly displayed.
- When recording telephone conversations, we recommend informing the other party that the conversation is being recorded.
- Be sure to comply with applicable local regulations (laws, ordinances, guidelines, etc.) regarding telephone conversation recording.

Recording Conversations

1. Press REC during a conversation.
   - "**`Remaining Time`**" and the approximate recording time available are displayed briefly, then "**`Conf Recording`**" and the approximate length of the call are displayed.
2. To stop recording, press STOP .
   - Recording stops automatically when **[ ]** is pressed.

Recording Voice Memos

You can record a voice memo while the unit is in standby mode.

1. Press **[MENU]** while the unit is in standby mode.
2. Select "**`Voice Memo`**", then press REC .
   - "**`Remaining Time`**" and the approximate recording time available are displayed briefly, then "**`Memo Recording`**" and the approximate length of the call are displayed.
3. To stop recording, press STOP .
   - Recording stops automatically when a call is received.

Playing Back Recordings

1. Press PLAY .
2. Press **[∧]** or **[∨]** repeatedly to scroll through the list of recordings, then press SELECT .
   - Recorded conversations are displayed as plus the date and time of the recording. Voice memos are displayed as plus the date and time of the recording.
3. Press PLAY .
   - Press **[BACK]** to stop playback.
   - If the selected recording is less than 1 second long, "**`Unable To Use`**" is displayed and the recording cannot be played back.
4. Press **[MENU]** to exit.

Note

- Recordings are saved in PCM format and can be played back on a computer using Windows Media® Player or QuickTime®. Recordings are stored on the SD memory card in the following folder: "\\PRIVATE\MEIGROUP\PCC\IPSP".

- When accessed by a computer, recordings are displayed as "**REC**" plus a 3 digit number (000–100; the lowest available number is used when a file is saved on the SD memory card). The file extension is "**.WAV**".
  Example: "**REC001.WAV**"
- If the names of files or folders are changed using a computer, the recordings cannot be played back using the unit.

Features Available During Playback

The following features are available during playback.

Feature	Operation
Volume control	Press [^] or [v]
Play next recording	Press **[#]** ([⏭]) then PLAY
Play current recording again	Press [*] ([⏮]) then PLAY
Play previous recording	Press [*] ([⏮]) 2 times, then PLAY
Fast forward	Press ⏩ for 4× speed Press ⏩ again for 60× speed Press PLAY for playback
Rewind	Press ⏪ for 4× speed Press ⏪ again for 60× speed Press PLAY for playback
Pause	Press **[0]** ([⏸]) Press PLAY to resume playback
Erase current recording	Press ERASE , then YES .

Erasing Recordings

Erasing 1 Recording

1. Press PLAY .
2. Press [^] or [v] repeatedly to scroll through the list of recordings, then press ERASE .
3. Press YES .
   - "**Erased**" is displayed.
4. Press **[MENU]**.

Note

- To erase a recording while listening to it, press ERASE , then press YES .

Erasing All Recordings

1. Press PLAY .
2. Press ERASE .
3. Press YES .
   - "**All Erased**" is displayed.

Using a Portable Station (PS)

By connecting the unit to a PS, you can use the unit's microphones and speaker for calls made or received with the PS.

Compatible Portable Stations (as of January, 2011)

– KX-TD7680 – KX-TD7685
– KX-TD7690 – KX-TD7695
– KX-TD7684 – KX-TD7696
– KX-TD7694 – KX-TD7896
– KX-WT125 – KX-WT126

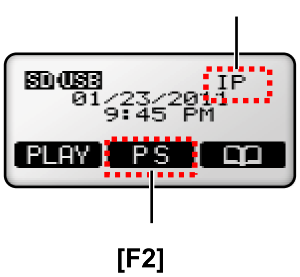
Line Selection (IP + PS)

1. Press **[MENU]**.
2. Press LINE .
   - When the operation mode is set to "**`USB Audio`**", LINE is not displayed and you cannot select a line type.
3. Select "**`IP + PS`**".
   - When "**`Line Selection`**" is set to "**`IP + PS`**", TEL calls cannot be made or received.
4. Press SAVE .
5. Press **[MENU]**.

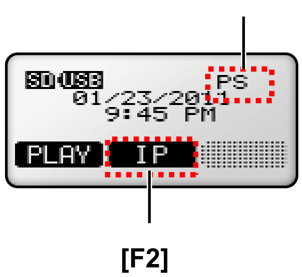
Using a PS

1. Confirm that "**`PS`**" is shown in the upper-right corner of the display.
   - You can change the selected line by pressing the center function button (**[F2]**).

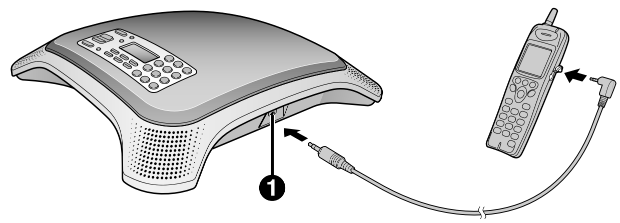
The IP line is selected.



The PS line is selected.



   - If you cannot select "**`PS`**", change the "**`Line Selection`**" setting to "**`IP + PS`**" (explained on this page).
2. Make or answer a call using the PS.
3. Connect the included PS Cable to the **[AUDIO IN/OUT]** jack of the unit (❶), and to the headset jack of the PS.



   - Make sure the plug of the PS Cable is inserted fully into the unit and PS, otherwise sound may not be heard, or echoing and feedback may occur.
   - Place the PS as far away from the unit as possible.
4. Press **[🕪]** to begin using the unit's microphones and speaker.
   - Press **[🕪]** again to turn off the microphones and speaker.
5. Use the PS to end the call.
6. Turn off the unit's microphones and speaker by pressing **[🕪]**.

Note

- For best performance, we recommend setting the receiver volume of the PS as follows:
  – 4-volume level models: level 3
  – 6-volume level models: level 4
- For best performance, do not change the speaker (receiver) volume of the PS frequently during a call. Echoing or feedback may occur.
- The microphone and receiver of the PS cannot be used while the PS is connected to the unit.
- This feature will not function when a mobile phone or incompatible PS is connected to the unit.

Using a Computer

By connecting the unit to a computer, you can use the unit's microphones and speaker for calls made or received with the computer using your preferred IP phone software. You can connect using the included USB Cable or PC Cable.

Connecting Using the USB Cable

Changing the Operation Mode

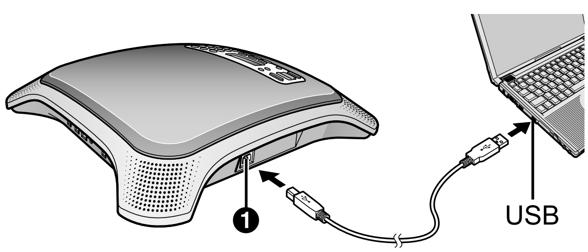
1. Press **[MENU]**.
2. Select "`Operation Mode`", then press **SELECT** .
3. Select "`USB Audio`".
4. Press **SAVE** .
5. Press **[MENU]**.

Notice

- If this setting is changed, the unit must be restarted before the new settings becomes effective (see page 47).
- When the operation mode is set to "`USB Audio`", IP, TEL, PS or PC calls cannot be made or received. Also, Conferencing Phone Manager cannot be operated.

Connecting to a Computer

1. Connect the included USB Cable to the USB port (❶) of the unit, and to the USB port of the computer.

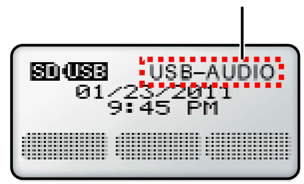


   - The new hardware wizard is displayed on the computer. If the new hardware wizard is not displayed automatically, set the operation mode to "`IP-PBX`" or "`Peer to Peer`" (see page 29).
2. Select **[Install the software automatically (Recommended)]** and then click **[Next]**.
   - A dialog may be displayed that indicates the software has not passed Windows logo testing. This is normal. The software will not cause any difficulties with your operating system. Click **[Continue Anyway]** to proceed with installation.
3. Click **[Finish]**.

Using a Computer

1. Confirm that "`USB-AUDIO`" is shown in the upper-right corner of the display.

"`USB Audio`" mode is selected.



   - If "`USB-AUDIO`" is not shown on the display, change the operation mode to "`USB Audio`" (explained on this page).
2. Press **[ ]** to begin using the unit's microphones and speaker.
   - Press **[ ]** again to turn off the microphones and speaker.
3. Use the desired computer software to make or answer a call.
4. Use the software to end the call.
5. Turn off the unit's microphones and speaker by pressing **[ ]**.

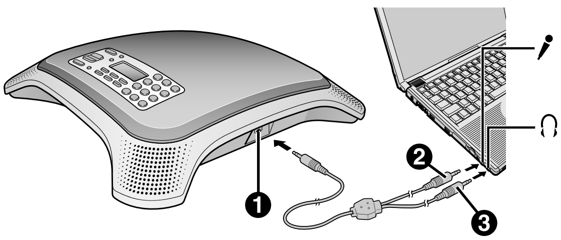
Note

- Change the settings of the following parameters of the IP phone software while on a call.
  - AGC (Automatic Gain Control): OFF
  - Echo canceling: OFF
  - Volume: Middle
- When changing the speaker volume while on a call, change the unit's volume setting. Keep the volume of the IP phone software on the middle level.
- If feedback or echoing occurs while on a call, decrease the speaker volume of the computer or the IP phone software.
- Leave at least 50 cm (about 20 in.) of space between the unit and the computer.

Connecting Using the PC Cable

Connecting a Computer

Connect the included PC Cable to the **[AUDIO IN/OUT]** jack of the unit (❶), and to the microphone jack (red plug; ❷) and headphone jack (green plug; ❸) of the computer.



Note

- The microphone and speaker of the computer cannot be used while the computer is connected to the unit.

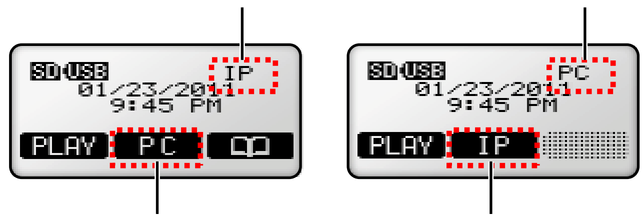
Line Selection (IP + PC)

1. Press **[MENU]**.
2. Press LINE .
   - When the operation mode is set to "`USB Audio`", LINE is not displayed and you cannot select a line type.
3. Select "`IP + PC`".
   - When "`Line Selection`" is set to "`IP + PC`", TEL calls cannot be made or received.
4. Press SAVE .
5. Press **[MENU]**.

Using a Computer

1. Confirm that "`PC`" is shown in the upper-right corner of the display.
   - You can change the selected line by pressing the center function button (**[F2]**).

The IP line is selected. The PC line is selected.



[F2] **[F2]**

   - If you cannot select "`PC`", change the "`Line Selection`" setting to "`IP + PC`" (explained on this page).
2. Press **[ ]** to begin using the unit's microphones and speaker.
   - Press **[ ]** again to turn off the microphones and speaker.
3. Use the desired computer software to make or answer a call.
4. Use the computer to end the call.
5. Turn off the unit's microphones and speaker by pressing **[ ]**.

Note

- Leave at least 50 cm (about 20 in.) of space between the unit and the computer.

Changing and Confirming Settings

Settings can be programmed and confirmed from the front panel. To use Conferencing Phone Manager (found on the included CD-ROM) to program the unit from a computer, see the Operating Instructions for Conferencing Phone Manager for more details.

Changing Settings

Using the Front Panel

1. Press **[MENU]**.
2. Press **[∧]** or **[∨]** to select the desired programming category, then press SELECT.
3. Press **[∧]** or **[∨]** to select the desired parameter, then press SELECT.
4. Adjust the settings as desired.
   - To cancel without changing any settings, press **[MENU]**.
5. Press SAVE.
6. Press **[MENU]**.

Note

- When programming the unit from the front panel, the current setting is indicated on the display by "✓".
- To return to the previous screen, press **[BACK]**.
- If you do not press any buttons for 1 minute, the unit will return to standby mode.
- The following parameters cannot be programmed, and are therefore not displayed, when the unit is in use.
  – `Operation Mode`
  – `Language`
  – `Date & Time`
  – `Time Format`
  – `Ringer Volume`
  – `Ringtone`
  – `IP Network Settings` (all parameters)
  – `Protocol Settings` (all parameters)
  – `VoIP Settings` (all parameters)
  – `QoS Settings` (all parameters)

  Additionally, the following parameters cannot be programmed, and are therefore not displayed, when the TEL, PS, or PC line is in use.
  – `Line Selection`
  – `TEL Settings` (all parameters)

Using Conferencing Phone Manager

To use Conferencing Phone Manager (found on the included CD-ROM) to program the unit from a computer, see the Operating Instructions for Conferencing Phone Manager for more details.

Restarting the Unit

After changing the following parameters, the unit must be restarted in order for the new settings to take effect. (Data is not erased when the unit is restarted.)

– Operation Mode (page 51)
– IP Address Mode (page 54)
– IP Address (page 55)
– Subnet Mask (page 55)
– Default Gateway (page 55)
– SIP Ext. No. (page 55)
– SIP Authentication ID (page 56)
– SIP Password (page 56)
– SIP User Domain Name (page 56)
– SIP Proxy Server IP Address (page 56)
– SIP Proxy Server Port Number (page 57)
– SIP Registrar IP Address (page 57)
– SIP Registrar Port Number (page 57)
– Hold Method (page 57)
– Panasonic PBX Compatibility (page 58)
– SIP Signaling Port Number (page 58)
– VLAN ID (page 59)
– VLAN Priority (page 59)
– SIP ToS Field (page 60)
– RTP ToS Field (page 60)

1. Press **[MENU]**.
2. Select "`System Options`", then press SELECT.
3. Select "`Reboot`", then press SELECT.
4. Select "`Yes`", then press SELECT.

Note

- To use Conferencing Phone Manager to restart the unit, see the Operating Instructions for Conferencing Phone Manager for more details.
- You can also restart the unit by turning it off and on again:
  – **When using the AC adaptor:** Disconnect the AC adaptor from the AC outlet, then connect it again.

– **When using PoE:** Disconnect the LAN cable from the unit, then connect it again.

Parameter List

The following is a list of all programmable parameters. See the references listed here for information about each parameter.

Programming Category	Parameter	Reference
`SIP Ext. No.` (IP-PBX mode only)		page 51
`Show IP Address`		page 51
`Operation Mode`		page 51
`Basic Settings`	`Language`	page 51
	`Date & Time`	page 52
	`Time Format`	page 52
	`Ringer Volume`	page 52
	`Ringtone`	page 52
	`LCD Contrast`	page 52
	`Key Tones`	page 53
	`Call Waiting Tone Volume`	page 53
	`Call Waiting`	page 53
`Line Selection`		page 53
`TEL Settings`	`Dial Mode`	page 53
	`Flash Time`	page 54
	`Pause Time`	page 54
	`TEL Line Level Reduction`	page 54
`IP Network Settings`	`IP Address Mode`	page 54
	`IP Address`	page 55
	`Subnet Mask`	page 55
	`Default Gateway`	page 55
`Protocol Settings`	`SIP Ext. No.`	page 55
	`SIP Authentication ID`	page 56
	`SIP Password`	page 56
	`SIP User Domain Name`	page 56
	`SIP Proxy Server IP Address`	page 56
	`SIP Proxy Server Port Number`	page 57

Programming Category	Parameter	Reference
	SIP Registrar IP Address	page 57
	SIP Registrar Port Number	page 57
	Hold Method	page 57
	Panasonic PBX Compatibility	page 58
VoIP Settings	Preferred CODEC	page 58
	RTP Packet Size	page 58
	SIP Signaling Port Number	page 58
	RTP Port Number (Minimum)	page 59
	RTP Port Number (Maximum)	page 59
	DTMF Type	page 59
QoS Settings	VLAN ID	page 59
	VLAN Priority	page 59
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Parameters

When programming the unit from the front panel, the current setting is indicated on the display by "✓".

SIP Ext. No.

Description
Allows you to confirm the unit's SIP extension number. This parameter is only available when the operation mode is set to "`IP-PBX`". To change the SIP extension number, see page 28.

Show IP Address

Description	Available Settings	Default Setting
Allows you to confirm the unit's IP address. To change the IP address, see page 27.	—	`192.168.0.2`

Operation Mode

Description	Available Settings	Default Setting
The unit can be operated as a SIP extension of the PBX, a peer to peer IP conferencing telephone, or a computer's USB audio device. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	☐ `IP-PBX` ☐ `Peer to Peer` ☐ `USB Audio`	`IP-PBX`

Basic Settings

Language

Description	Available Settings	Default Setting
Determines the display language. Use the following procedure to change the language. **1.** Press **[MENU]**. **2.** Press **[⌃]** or **[⌄]** repeatedly to select "`Basic Settings`". **3.** Press **[ENTER]** two times. **4.** Press **[⌃]** or **[⌄]** repeatedly to select the desired language. **5.** Press **[ENTER]**. **6.** Press **[MENU]**.	☐ `English(US)` ☐ `Español` ☐ `Français` ☐ `Français(C)` ☐ `Italiano` ☐ `Nederlands` ☐ `Português` ☐ `Deutsch` ☐ `English(UK)`	`English(US)`

Date & Time

Description	Available Settings	Default Setting
The date and time are shown on the display in standby mode, and are displayed when playing conversations that were recorded to an SD memory card. • Using the keypad, enter 2 digits each for the year, month, day of the month, hour (24-hour format), and minute. Example: To enter "Jan. 23, 2011, 7:45 PM", press **[1101231945]**. • To correct a mistake, press [<] to move the cursor as needed, then enter the correct number.	Year, month, day of the month, hour (24-hour format), minute	Jan. 1, 2011, 12:00 AM Date format is determined by the display language (see page 51). English(US): MM/DD/YYYY Español: DD/MM/YYYY Français: DD/MM/YYYY Français(C): YYYY-MM-DD Italiano: DD/MM/YYYY Nederlands: DD-MM-YYYY Português: DD-MM-YYYY Deutsch: DD.MM.YYYY English(UK): DD/MM/YYYY

Time Format

Description	Available Settings	Default Setting
Determines the time format.	☐ `12-hour` ☐ `24-hour`	`12-hour`

Ringer Volume

Description	Available Settings	Default Setting
Determines the ringer volume.	4 levels including "`Off`"	High

Ringtone

Description	Available Settings	Default Setting
Determines the ringtone.	☐ `Tone Pattern 1` ☐ `Tone Pattern 2` ☐ `Tone Pattern 3`	`Tone Pattern 1`

LCD Contrast

Description	Available Settings	Default Setting
Determines the LCD contrast.	6 levels	Level 3

Key Tones

Description	Available Settings	Default Setting
Determines whether tones are heard when the unit's buttons are pressed.	☐ **On** ☐ **Off**	**On**

Call Waiting Tone Volume

Description	Available Settings	Default Setting
Determines the volume of the call waiting tone heard when a second call is received.	☐ **High** ☐ **Low**	**High**

Call Waiting

Description	Available Settings	Default Setting
Determines whether a call can be received when you are already on another call.	☐ **Enable** ☐ **Disable**	**Enable**

Line Selection

Description	Available Settings	Default Setting
Determines which lines can be used to make and receive calls.	☐ **IP + TEL** ☐ **IP + PC** ☐ **IP + PS**	**IP + TEL**

TEL Settings

Dial Mode

Description	Available Settings	Default Setting
Determines the dial mode used for the TEL line. • Set this parameter to match the specification of the TEL line.	☐ **Pulse** ☐ **Tone**	**Tone**

Flash Time

Description	Available Settings	Default Setting
Determines the flash time. • Set this parameter to match the specification of the TEL line.	☐ 900 ms ☐ 700 ms ☐ 600 ms ☐ 400 ms ☐ 300 ms ☐ 250 ms ☐ 200 ms ☐ 160 ms ☐ 110 ms ☐ 100 ms ☐ 90 ms ☐ 80 ms	700 ms

Pause Time

Description	Available Settings	Default Setting
Determines the length of the dialing pause inserted when **[REDIAL/PAUSE]** is pressed.	☐ 3 s ☐ 5 s	3 s

TEL Line Level Reduction

Description	Available Settings	Default Setting
Determines whether the incoming TEL line signal level is reduced. Set this feature to "On" if the volume of your voice heard from the speaker is too loud. • When set to "On", the other party's voice will be slightly quieter. • We recommend setting this feature to "On" when establishing conference calls that use the TEL line.	☐ On ☐ Off	Off

IP Network Settings

Consult your system administrator for the appropriate settings.

IP Address Mode

Description	Available Settings	Default Setting
Determines whether the unit's IP address is assigned automatically (DHCP) or manually (static). • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	☐ DHCP ☐ Static	Static

IP Address

Description	Available Settings	Default Setting
Determines the unit's IP address. • This parameter is only available when "**`IP Address Mode`**" is set to "**`Static`**". • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	—	**`192.168.0.2`**

Subnet Mask

Description	Available Settings	Default Setting
Set this parameter to match the IP address of the IP network's subnet mask. • This parameter is only available when "**`IP Address Mode`**" is set to "**`Static`**". • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	—	**`255.255.255.0`**

Default Gateway

Description	Available Settings	Default Setting
Set this parameter to match the IP address of the IP network's default gateway. • This parameter is only available when "**`IP Address Mode`**" is set to "**`Static`**". • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	—	**`0.0.0.0`**

Protocol Settings

Consult your system administrator for the appropriate settings.

SIP Ext. No.

Description	Available Settings	Default Setting
Set this parameter to match the extension number assigned to the unit through PBX programming. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	Max. 32 characters	—

SIP Authentication ID

Description	Available Settings	Default Setting
Set this parameter to match the SIP authentication ID assigned to the unit through PBX programming. If this field is left empty, the value set for "**`SIP Ext. No.`**" will be used as the SIP authentication ID. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	Max. 32 characters	—

SIP Password

Description	Available Settings	Default Setting
Set this parameter to match the password assigned to the unit through PBX programming. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	Max. 32 characters	—

SIP User Domain Name

Description	Available Settings	Default Setting
Set this parameter to match the IP address or domain name of the IP-PBX. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	Max. 64 characters	—

SIP Proxy Server IP Address

Description	Available Settings	Default Setting
Set this parameter to match the IP address of the IP-PBX. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	—	`0.0.0.0`

SIP Proxy Server Port Number

Description	Available Settings	Default Setting
Set this parameter to match the SIP port number of the IP-PBX or SIP proxy server. The communication protocol used is UDP. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	1024–50999	5060

SIP Registrar IP Address

Description	Available Settings	Default Setting
Set this parameter to match the IP address of the IP-PBX. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	—	0.0.0.0

SIP Registrar Port Number

Description	Available Settings	Default Setting
Set this parameter to match the SIP port number of the IP-PBX or SIP registrar server. The communication protocol used is UDP. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	1024–50999	5060

Hold Method

Description	Available Settings	Default Setting
Determines the hold protocol used for the IP line. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	☐ RFC2543 ☐ RFC3264	RFC3264

Panasonic PBX Compatibility

Description	Available Settings	Default Setting
Determines whether the unit can be connected to a Panasonic PBX. • Set this setting to "On" when connecting to a Panasonic PBX. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	☐ On ☐ Off	On

VoIP Settings

Preferred CODEC

Description	Available Settings	Default Setting
Determines the preferred codec for IP calls.	☐ G.722 ☐ G.711μ-law ☐ G.711A-law ☐ G.729a	G.722

RTP Packet Size

Description	Available Settings	Default Setting
Determines the size of outgoing RTP packets.	☐ 20 ms ☐ 30 ms ☐ 40 ms ☐ 50 ms ☐ 60 ms	20 ms

SIP Signaling Port Number

Description	Available Settings	Default Setting
Determines the port number used for SIP signaling. The communication protocol used is UDP. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	1024–50999	5060

RTP Port Number (Minimum)

Description	Available Settings	Default Setting
Determines the lowest port number used for RTP. The communication protocol used is UDP. • Set this parameter to a value lesser than the setting for "**RTP Port Number (Maximum)**".	51000–51998 (even values only)	51000

RTP Port Number (Maximum)

Description	Available Settings	Default Setting
Determines the highest port number used for RTP. The communication protocol used is UDP. • Set this parameter to a value greater than the setting for "**RTP Port Number (Minimum)**".	51002–52000 (even values only)	52000

DTMF Type

Description	Available Settings	Default Setting
Determines the type of DTMF signal sent during a call.	☐ **In-band** ☐ **Out-of-band**	**Out-of-band**

QoS Settings

VLAN ID

Description	Available Settings	Default Setting
Determines the VLAN ID. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	0001–4095	4095

VLAN Priority

Description	Available Settings	Default Setting
Determines the VLAN priority. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	0–7	0

SIP ToS Field

Description	Available Settings	Default Setting
Determines the value of the SIP ToS Field. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	8-bit binary code	00000000

RTP ToS Field

Description	Available Settings	Default Setting
Determines the value of the RTP ToS Field. • If this setting is changed, the unit must be restarted before the new setting becomes effective (see page 47).	8-bit binary code	00000000

System Status Confirmation

Description
The settings for the following parameters can be displayed for confirmation. – Software Version – IP Address – Subnet Mask – Default Gateway – DHCP Server – MAC Address

System Options

Erase All Call Log Data

Description
Allows you to erase all entries in the redial list.

Erase All Phonebook Data

Description
Allows you to erase all entries in the phonebook.

Reset System Data

Description
Allows you to reset all system data (i.e., the parameters described in this section) to the factory default settings. • After executing this feature, the unit will restart automatically. • The following data and settings *are not erased* when this feature is executed. – "**Date & Time**" setting (note that the "**Time Format**" setting is reset) – Phonebook entries – Redial list – SD memory card recordings

Reset All Data

Description
Allows you to reset all system data (i.e., the parameters described in this section), user data, and all settings. The unit is returned to its factory default state when this feature is executed. • After executing this feature, the unit will restart automatically. • The following data and settings *are not erased* when this feature is executed. – SD memory card recordings • The following data and settings **are erased** when this feature is executed. – "**Date & Time**" setting – Phonebook entries – Redial list

Reboot

Description
Allows you to restart the unit. • No data is erased when this feature is executed.

Entering Characters

The dial keys can be used to enter characters on the unit.
Available characters depend on the type of entry being made, as follows:

- Making a call after entering a SIP extension number or SIP URI (see page 34)
- Storing a SIP extension number or SIP URI in the phonebook (see page 40)
- Storing a name in the phonebook (see page 40)
- Changing SIP settings ("**SIP Ext. No.**", "**SIP Authentication ID**", "**SIP Password**" or "**SIP User Domain Name**") (see page 28)

Character Tables

SIP Extension Number or SIP URI Entry When Dialing or Storing in Phonebook

Dial Key	Alphabet Entry Mode	
	Uppercase	Lowercase
0	( ) 0	
1	. ! ' * - _ ~ 1	
2	A B C 2	a b c 2
3	D E F 3	d e f 3
4	G H I 4	g h i 4
5	J K L 5	j k l 5
6	M N O 6	m n o 6
7	P Q R S 7	p q r s 7
8	T U V 8	t u v 8
9	W X Y Z 9	w x y z 9
*	Changes between uppercase and lowercase character entry.	
#	@ # $ & + , =	

Note

- "@" can only be used when entering a SIP URI.
- A period can be entered by pressing **[1]**, when the character entry mode is alphabet entry mode.

Phonebook Name Entry

Dial Key	Alphabet Entry Mode	Extended Character Entry Mode
0	( ) Space < > [ ] { } 0	( ) Space < > [ ] { } 0
1	. ! ' ∗ - _ ~ " % \ ^ ` | 1	. ! ' ∗ - _ ~ " % \ ^ ` | 1
2	A B C 2	A B C À Á Â Ã Ä Å Æ Ç 2
	a b c 2	a b c à á â ã ä å æ ç 2
3	D E F 3	D E F È É Ê Ë Ẽ 3
	d e f 3	d e f è é ê ë ẽ 3
4	G H I 4	G H I Ğ Ì Í Î Ï Ĩ İ Ĭ 4
	g h i 4	g h i ğ ì í î ï ĩ ı ĭ 4
5	J K L 5	J K L 5
	j k l 5	j k l 5
6	M N O 6	M N O Ñ Ò Ó Ô Õ Ö Ø 6
	m n o 6	m n o ñ ò ó ô õ ö ø 6
7	P Q R S 7	P Q R S Ş ß 7
	p q r s 7	p q r s ş ß 7
8	T U V 8	T U V Ù Ú Û Ü Ũ 8
	t u v 8	t u v ù ú û ü ũ 8
9	W X Y Z 9	W X Y Z Ŵ Ŷ 9
	w x y z 9	w x y z ŵ ŷ 9
∗	Changes between uppercase and lowercase character entry.	Changes between uppercase and lowercase character entry.
#	@ # $ & + , = / : ; ?	@ # $ & + , = / : ; ?

Note

- A space counts as one character.
- A period can be entered by pressing **[1]**, when the character entry mode is alphabet entry mode or extended character entry mode.

SIP Settings

Dial Key	Alphabet Entry Mode	
	Uppercase	Lowercase
0	( ) 0	
1	. ! ' * - _ ~ 1	
2	A B C 2	a b c 2
3	D E F 3	d e f 3
4	G H I 4	g h i 4
5	J K L 5	j k l 5
6	M N O 6	m n o 6
7	P Q R S 7	p q r s 7
8	T U V 8	t u v 8
9	W X Y Z 9	w x y z 9
*	Changes between uppercase and lowercase character entry.	
#	# $ & + , =	

Note

- A period can be entered by pressing **[1]**, when the character entry mode is alphabet entry mode.

Erasing Data

The following features allow you to erase system data and user data stored in the unit.

Erase All Call Log Data
Allows you to erase all entries in the redial list.

Erase All Phonebook Data
Allows you to erase all entries in the phonebook.

Reset System Data
Allows you to reset all system data to the factory default settings.

- The following data and settings *are not erased* when this feature is executed.
  - "**Date & Time**" setting (note that the "**Time Format**" setting is reset)
  - Phonebook entries
  - Redial list
  - SD memory card recordings

Reset All Data
Allows you to reset all system data, user data, and all settings. The unit is returned to its factory default state when this feature is executed.

- The following data and settings *are not erased* when this feature is executed.
  - SD memory card recordings
- The following data and settings **are erased** when this feature is executed.
  - "**Date & Time**" setting
  - Phonebook entries
  - Redial list

Note

- These features do not erase the contents of the SD memory card. To erase the SD memory card, see page 43.

Erasing the Redial List

1. Press **[MENU]**.
2. Select "**System Options**", then press SELECT .
3. Select "**Erase All Call Log Data**", then press SELECT .
4. Select "**Yes**", then press SELECT .
5. Press **[MENU]**.

Erasing the Phonebook

1. Press **[MENU]**.
2. Select "**System Options**", then press SELECT .
3. Select "**Erase All Phonebook Data**", then press SELECT .
4. Select "**Yes**", then press SELECT .
5. Press **[MENU]**.

Resetting System Data

1. Press **[MENU]**.
2. Select "**System Options**", then press SELECT .
3. Select "**Reset System Data**", then press SELECT .
4. Select "**Yes**", then press SELECT .
   - The unit will restart automatically.

Resetting All Data

Notice

- The unit is returned to its factory default state when this feature is executed. All user data (except for the contents of the SD memory card) will be erased.

1. Press **[MENU]**.
2. Select "**System Options**", then press SELECT .
3. Select "**Reset All Data**", then press SELECT .
4. Select "**Yes**", then press SELECT .
   - The unit will restart automatically.

Troubleshooting

If you are experiencing trouble, refer to the information in this section. Before troubleshooting, confirm all connections (see page 25) and confirm that the AC outlet or PoE device to which the unit is connected to is receiving power.

General Use

Issue	Possible Cause & Solution	Reference
The display is blank.	• The unit is not receiving power. → The unit is not designed to function when there is a power failure. Make sure that the unit is connected to the AC outlet and receiving power. If using PoE, confirm that the device supplying PoE is receiving power and that the LAN cable is properly connected.	page 25
The unit is not performing properly.	• Cables or cords are not connected properly. → Check all connections.	page 25
	• An error has occurred. → Reset the unit. Disconnect the unit from the AC outlet, wait 10 seconds, then connect the AC adaptor again. If using PoE, disconnect the LAN cable, wait 10 seconds, then connect the LAN cable again.	page 25

Making and Receiving Calls

Issue	Possible Cause & Solution	Reference
I cannot make IP calls.	• The IP line is not selected. → Before dialing, confirm that "**IP**" is shown in the upper-right corner of the display. Press the center function button (**[F2]**) to change the line if necessary.	page 33
	• The IP address was entered incorrectly. → Confirm that you have entered the IP address of the other party correctly.	page 34
	• The "**Operation Mode**" setting is not correct. → Change the setting to "**Peer to Peer**" if you want to make or receive peer to peer IP calls (i.e., calls made by specifying the called party's IP address). → Change the setting to "**IP-PBX**" if you want to make or receive intercom and outside calls as a SIP extension (i.e., calls made by specifying the called party's SIP extension number or SIP URI). → When the setting is "**USB Audio**", you cannot make or receive IP calls. Change the setting to "**Peer to Peer**" or "**IP-PBX**".	page 29
	• Consult your system administrator.	—

Issue	Possible Cause & Solution	Reference
I cannot make TEL calls.	• The TEL line is not selected. → Before dialing, confirm that "**TEL**" is shown in the upper-right corner of the display. Press the center function button (**[F2]**) to change the line if necessary.	page 33
	• The unit is not set to make TEL calls. → Make sure the "**Line Selection**" setting is set to "**IP + TEL**".	page 26
	• The operation mode is set to "**USB Audio**". → When the operation mode is set to "**USB Audio**", you cannot make or receive IP, TEL, PS or PC calls. Change the setting to "**Peer to Peer**" or "**IP-PBX**".	page 29
	• The dial mode setting is incorrect. → Make sure that the dial mode matches the type of telephone service you have (i.e., tone or pulse).	page 26
I cannot receive IP calls.	• The "**Operation Mode**" setting is not correct. → Change the setting to "**Peer to Peer**" if you want to make or receive peer to peer IP calls (i.e., calls made by specifying the called party's IP address). → Change the setting to "**IP-PBX**" if you want to make or receive intercom and outside calls as a SIP extension (i.e., calls made by specifying the called party's SIP extension number or SIP URI). → When the setting is "**USB Audio**", you cannot make or receive IP calls. Change the setting to "**Peer to Peer**" or "**IP-PBX**".	page 29
I cannot receive TEL calls.	• The unit is not set to receive TEL calls. → Make sure the "**Line Selection**" setting is set to "**IP + TEL**".	page 26
	• The operation mode is set to "**USB Audio**". → When the operation mode is set to "**USB Audio**", you cannot make or receive IP, TEL, PS or PC calls. Change the setting to "**Peer to Peer**" or "**IP-PBX**".	page 29

Issue	Possible Cause & Solution	Reference
I cannot use the unit for PS or PC calls.	• The Portable Station (PS) or computer is not connected correctly. → Confirm you are using the correct cable to connect to the device, and that the cable is connected properly.	page 44 page 45
	• The operation mode is set to "`USB Audio`". → When the operation mode is set to "`USB Audio`", you cannot make or receive IP, TEL, PS or PC calls. Change the setting to "`Peer to Peer`" or "`IP-PBX`".	page 29
	• The PS or PC line is not selected. → Confirm that "`PS`" or "`PC`" is shown in the upper-right corner of the display. Press the center function button (**[F2]**) to change the line if necessary. → Make sure the "`Line Selection`" setting is set to "`IP + PS`" (for PS calls) or "`IP + PC`" (for PC calls).	page 44 page 45
	• The PS or computer is not able to make or receive calls. → Disconnect the audio cable from the device and confirm that the device is able to make or receive calls. If the device cannot make calls, you cannot use the unit for PS or PC calls. Refer to the operating instructions for the device.	—
I cannot redial by pressing **[REDIAL/PAUSE]**.	• The number you are trying to redial is too long. → If the dialed number contains too many characters, it cannot be redialed correctly. The maximum for each type of number is as follows: – TEL line: 128 digits – IP line (SIP extension number): 32 characters – IP line (SIP URI): 97 characters (excluding "sip:")	—
	• You pressed **[REDIAL/PAUSE]** after you began dialing. → If you press **[REDIAL/PAUSE]** after you begin dialing a phone number, the button functions as the pause button. To redial the last number dialed, press **[ ]** then **[REDIAL/PAUSE]**. To redial one of the last 10 phone numbers dialed, press **[REDIAL/PAUSE]**, press **[∧]** or **[∨]** to select the desired phone number, then press **[ ]**.	page 35
	• You are trying to redial a TEL line call, but the unit is not set to make TEL calls. → Make sure the "`Line Selection`" setting is set to "`IP + TEL`".	page 26
	• The "`Operation Mode`" setting is not the same as when the original call was made. (For example, you are trying to redial an IP call made in IP-PBX mode, but the unit is now in peer to peer mode.) → Change the setting.	page 29

Issue	Possible Cause & Solution	Reference
I cannot make long distance calls.	• Your telephone service does not allow you to make long distance calls. → Make sure that you have subscribed to your telephone company's long distance service.	—
	• If the unit is connected to a PBX, your extension may not be configured to make long distance phone calls. → Contact the PBX administrator.	—
The unit does not ring when a call is received.	• The ringer is turned off. → Press **[^]** while a call is being received, or change the ringer volume setting.	page 38
I cannot make a 3-party call using 2 IP lines in "**IP-PBX**" mode.	• The "**Panasonic PBX Compatibility**" setting is set to "**On**". → Change the setting to "**Off**".	page 58

Sound Quality

Issue	Possible Cause & Solution	Reference
The other party cannot hear my voice.	• The unit is muted. → If "**Mute**" is displayed, press **[🎙/C]** to turn off the mute feature.	page 39
	• Objects are obstructing the microphone. → Do not obstruct the unit or an External Wired MIC during calls. Keep your hands, as well as common objects such as folders, cups, and coffee pots away from the unit and the External Wired MIC during calls.	—

Issue	Possible Cause & Solution	Reference
Sound cuts out; I can hear myself through the speaker.	• The unit has not yet adapted to the environment for the current call. → Speak in turns with the other party at the beginning of a conversation. This allows the unit to adapt to its environment so that both parties can speak effectively.	—
	• If using a PS, you are pressing **[🔊]** before the PS call has started. → Start the call using the PS, then press **[🔊]**.	page 44
	• If using a PS, the PS Cable is not firmly connected to the PS. → Make sure the plug of the PS Cable is inserted fully into the PS, otherwise echoing and feedback may occur.	page 44
	• If using DSL service, a device connected between the unit and the telephone jack is causing interference. → Connect the unit directly to the telephone jack if possible, and/or consult your DSL service provider.	—
	• You are too far away from the microphone. → Try speaking closer to the microphone.	page 32
	• The environment is not suited to speakerphone calls. → Do not use the unit within 2 m (about 6.5 ft.) of projectors, air conditioning devices, fans, or other audible or electrical noise emitting devices. → If using the unit in a room with windows, close the curtains or blinds to prevent echoes. → Use the unit in a quiet environment.	—
	• The unit or an External Wired MIC was moved during a call. → Do not move the unit or an External Wired MIC while on a call.	—
	• Objects are obstructing the microphone. → Do not obstruct the unit or an External Wired MIC during calls. Keep your hands, as well as common objects such as folders, cups, and coffee pots away from the unit and the External Wired MIC during calls.	—
	• The other party is using a half-duplex speakerphone. → If the other party is using a half-duplex speakerphone, sound may cut out occasionally during calls. For best performance, the other party should use a full-duplex speakerphone.	—

SD Memory Cards

Issue	Possible Cause & Solution	Reference
I cannot record to the SD memory card.	• The SD memory card is not compatible with the unit. → Make sure that you are using a compatible SD memory card. (SDHC, miniSDHC, and microSDHC memory cards are not compatible with the unit.)	page 30
	• The SD memory card is not formatted or was not formatted properly. → Format SD memory cards for use with this unit in FAT format (FAT16) using a computer.	page 31
	• The SD memory card was not inserted properly. → Make sure that the SD memory card is inserted properly by gently pushing it securely but gently toward the back of the SD memory card slot.	page 30
	• The switch on the side of the SD memory card is in the "LOCK" position. → Remove the card, slide the switch to unlock the card, then insert the card again.	page 30

Phonebook

Issue	Possible Cause & Solution	Reference
I cannot add or edit entries to the phonebook.	• You are on a call or playing back recordings. → You cannot add or edit phonebook entries while on a call or while playing back recordings.	—
	• The "**Operation Mode**" setting is not correct. → Change the setting to "**Peer to Peer**" if you want to add or edit peer to peer IP call entries (i.e., entries stored by specifying the called party's IP address). → Change the setting to "**IP-PBX**" if you want to add or edit IP call entries that will be called when using the unit as a SIP extension (i.e., entries stored by specifying the called party's SIP extension number or SIP URI).	page 29
	• A call is being received. → The unit exits the phonebook automatically when a call is received. Add or edit the phonebook entry again once you have finished the call.	—
	• There are 100 entries in the phonebook. → The phonebook is full. Erase any unnecessary entries.	page 41

Issue	Possible Cause & Solution	Reference
I cannot call entries in the phonebook.	• You are trying to make a TEL line call, but the unit is not set to make TEL calls. → Make sure the "**Line Selection**" setting is set to "**IP + TEL**".	page 26
	• You are trying to make an IP line call, but the "**Operation Mode**" setting is not correct. → Change the setting to "**Peer to Peer**" if you want to make or receive peer to peer IP calls (i.e., calls made by specifying the called party's IP address). → Change the setting to "**IP-PBX**" if you want to make or receive intercom and outside calls as a SIP extension of the connected PBX.	page 29
The unit returns to standby mode while adding or editing phonebook entries.	• 1 minute has passed since you pressed a button. → If you pause for over 1 minute while adding or editing phonebook entries, the unit returns to standby mode.	—

Programming

Issue	Possible Cause & Solution	Reference
The unit returns to standby mode while programming the unit.	• 1 minute has passed since you pressed a button. → If you pause for over 1 minute while programming the unit, the unit returns to standby mode.	—
I cannot program the unit.	• You are on a call. → Program the unit once you have finished the call.	—
	• A call is being received. → The unit exits programming mode automatically when a call is received. Program the unit again once you have finished the call.	—
After I changed the settings, the changes do not take effect.	• The unit must be restarted before the new setting becomes effective. → Restart the unit.	page 47
The DTMF tone signal sent from the unit is not detected by the other party or PBX.	• The DTMF type differs from the type used by the other party or PBX. → Make sure that the "**DTMF Type**" setting matches the other party's or other PBX's setting.	page 59

Display Messages

Message	Possible Cause & Solution	Reference
Busy	• The called party is busy (displayed for IP calls only). → Try again later.	—

Message	Possible Cause & Solution	Reference
Reject Call	• The called party rejected your call (displayed for IP calls only). → Try again later.	—
Not Found	• The called party cannot be connected (displayed for IP calls only). → Confirm that you have entered the IP address, SIP extension, or SIP URI of the other party correctly. → Try again later.	—
Phonebook No Items Stored	• The phonebook contains no entries. → You must store entries in the phonebook before you can make a call from the phonebook.	page 40
Phonebook Error	• An error has occurred. → Press **[BACK]**, then erase all phonebook entries using the "**Erase All Phonebook Data**" feature.	page 65
Call Log Error	• An error has occurred. → Press **[BACK]**, then erase the redial list using the "**Erase All Call Log Data**" feature.	page 65
Please Wait	• An SD memory card was inserted. → Wait while the unit checks the card.	—
Format Error	• An error occurred while formatting the SD memory card. → Remove the SD memory card and use a different card.	—
Unable To Use	• The SD memory card is not compatible with the unit. → Make sure that you are using a compatible SD memory card. → SDHC, miniSDHC, and microSDHC memory cards are not compatible with the unit.	page 30
	• The SD memory card is not formatted in FAT format. → Use a FAT format SD memory card.	page 31
	• The selected recording is less than 1 second long. → Recordings less than 1 second long cannot be played back.	—
Memory Full	• The SD memory card cannot be used for recording because it is full. → Erase unneeded recordings.	page 43
Write Protected	• The switch on the side of the SD memory card is in the "LOCK" position. → Remove the card, slide the switch to unlock the card, then insert the card again.	page 30
System Data Err Clear Data?	• An error has occurred. → Press YES to reset all system data and restart the unit. Press NO to restart the unit without resetting any data.	—

Message	Possible Cause & Solution	Reference
No Connection To Analog Line	• You tried to make a call immediately after refusing a call. → Wait until "**TEL**" is shown on the display, then make the call.	—
No Connection To SIP Server	• The unit's network settings and/or SIP settings are incorrect. → Consult your system administrator.	—
No Connection To IP Network	• The LAN cable is not connected. → Check all connections.	page 25
	• The unit cannot receive IP settings from the DHCP server. → Consult your system administrator.	—
Hold Failure	• The call could not be put on hold. → Check the status of the other party (or IP-PBX). → Wait a while and try again.	—
Resume Failure	• The call could not be retrieved from hold status. → Check the status of the other party (or IP-PBX). → Wait a while and try again.	—

Cleaning the Unit

Clean the unit periodically with a soft, dry cloth.

Keep the following in mind when cleaning the unit.

- To avoid damaging the unit, disconnect the AC adaptor cord and all cables from the unit before cleaning.
- If the unit becomes particularly dirty, apply a light kitchen cleanser to a soft cloth, wring the cloth thoroughly, and wipe the unit. When finished, dry the unit with a soft, dry cloth.
- To avoid damage or discoloration, do not clean the unit with the following materials, or with cleaners containing the following materials.
  – Petroleum
  – Scouring powder
  – Alcohol
  – Paint thinner
  – Benzine
  – Wax
  – Hot water
  – Powdered soap
- When using chemical cleansers, follow the instructions on the label carefully.

Specifications

Item	Specification
Communication Lines	4 (IP, TEL, PS, PC) IP: Communication via IP network, available in peer to peer mode and IP-PBX mode TEL: Communication via telephone line PS: Communication via audio connection to a compatible Portable Station PC: Communication via audio connection to a computer
Maximum No. of Parties	3 – Main unit user + 2 IP calls – Main unit user + 1 IP call + 1 TEL call – Main unit user + 1 IP call + 1 PS call – Main unit user + 1 IP call + 1 PC call
VoIP Connection Method	SIP
VoIP Audio Codec	G.722, G.711 (μ-law/A-law), G.729a
LAN Interface	IEEE802.3/IEEE802.3u (10/100Base-TX) Straight/cross automatic crossover (Auto MDI/MDX)
IP Address Mode	Automatic (DHCP), manual (static)
DTMF	Out-of-band (RFC2833), In-band
Dial Mode	Tone, pulse
Speaker	1 (Output: 85 dB; Frequency range: 300 Hz–7000 Hz)
Built-in Microphone	4 locations, 8 total (Sensitivity area: about 3 m [about 10 ft.])
IP Network Interface Jack (LAN)	1 (RJ45)
Telephone Interface Jack (LINE)	1 (RJ11)
USB Port	1 (USB 2.0, Full speed)
SD Memory Card Slot	1 (32 MB–2 GB)
Audio Interface Jack (AUDIO IN/OUT)	1 (ø 3.5 mm, monaural)
External Wired MIC Jack (EXT MIC1, EXT MIC2)	2 (Modular jack)
Main Unit Dimensions	About 55 mm (H) × 275 mm (W) × 275 mm (D) *About 2 3/16 in. (H) × 10 1/16 in. (W) × 10 1/16 in. (D)*
Main Unit Mass	About 1300 g (2.87 lb.)
AC Adaptor	Input: AC 120 V, 60 Hz Output: DC 9 V 750 mA
PoE Interface	Compliant with IEEE802.3af
Power Consumption	Standby mode: about 4.8 W Talk mode: about 6.5 W

Item	Specification
Operating Environment	Temperature: 0 °C–40 °C (32 °F–104 °F) Humidity: Less than 90% (with no condensation) Ambient noise: Less than 50 dBA (recommended)

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When you ship the product
Carefully pack and send it prepaid, adequately insured and preferably in the original carton. Attach a postage-paid letter, detailing the symptom to the outside of the carton. DO NOT send the product to the Executive or Regional Sales offices. They are NOT equipped to make repairs.

Product service
For product service, ship the product to the address listed in the Limited Warranty. Consult your authorized Panasonic dealer for detailed instructions.

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http://www.panasonic.com/csd

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