Empowered by Innovation NEC

1. The Basics

2. The Components

3. Features

4. Specifications and Parts List

Aspire

Product Description

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This manual has been developed by NEC Unified Solutions, Inc. It is intended for the use of its customers and service personnel, and should be read in its entirety before attempting to install or program the system. Any comments or suggestions for improving this manual would be appreciated. Forward your remarks to:

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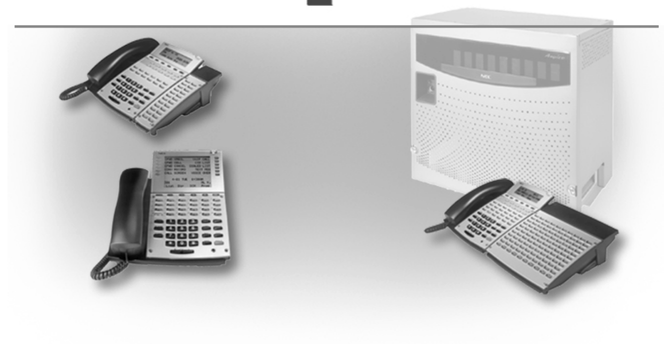
NEC

Aspire



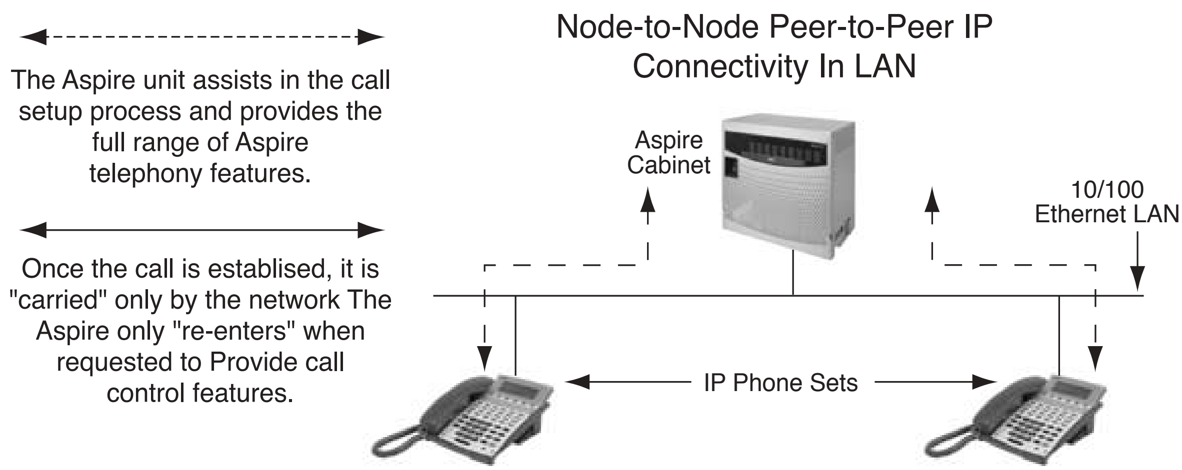
Aspire from NEC allows you to converge your voice and data network and enjoy the many advantages of Voice over Internet Protocol (VoIP) while enjoying the hundreds of features you've come to expect from traditional digital/analog switching. Aspire lets your organization benefit from the potential cost-saving advantages of IP even if you're not ready to migrate to 100% IP Telephony immediately. That's because Aspire gives you a choice: You can deploy traditional circuit-switched technology, VoIP or a combination, all from one system! You have the freedom to adopt VoIP when and where you need it.

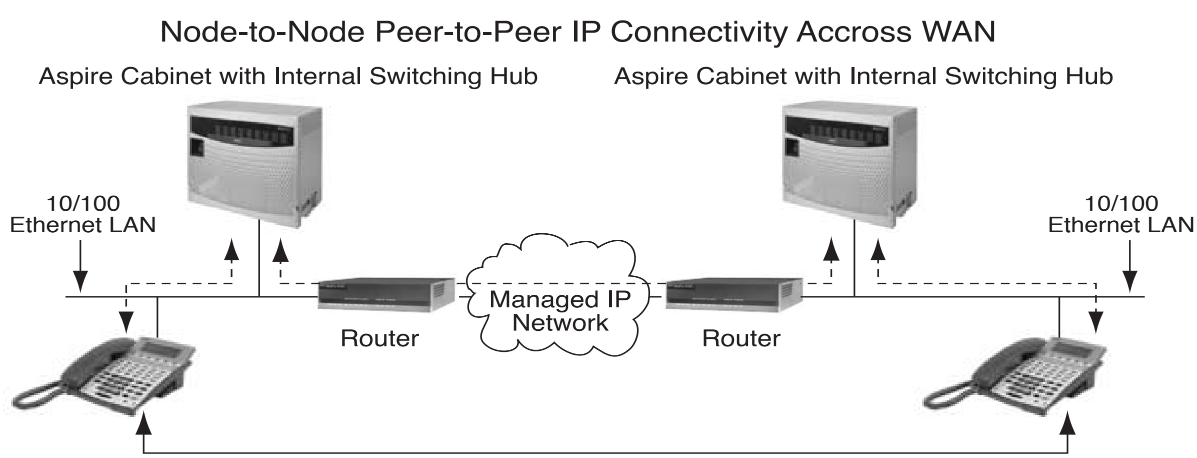
Peer-to-Peer Switching

"Peer-to-peer" switching means that the stations participating in a call are connected directly to each other through the IP network. The signals travel through the IP network but do not "go through" the switch as they do in traditional telephony. The fact that Aspire can function in and support a "hybrid" network with traditional digital/analog switching, IP/TDM/IP switching and pure peer-to-peer IP switching means that users can continue to utilize their existing equipment while they begin to phase in IP Telephony and lay the foundation for current and future networks.

Reduced Costs of Peer-to-Peer IP Connectivity

- Maintain one network rather than two
- Bypass the long distance carrier by sending voice calls over the data network
- Single cable termination to the desktop
- Reduced brick and mortar expenses by deploying main office features to remote personnel.





Elegant Sophisticated Design

Aspire's **Voice over IP (VoIP)** capability allows you to place voice calls over the data **network**. VoIP reduces long distance charges by using IP to connect multiple office locations and telecommuters. Remote workers also have access to main office features, such as voice mail, allowing offices to operate as a single unit.

Aspire Mail, an optional digitally linked in-switch voice mail, provides sophisticated features that save time and money. **Return call with Caller ID** saves caller ID information for inside- and outside-originated calls. **Answering Machine Emulation** is helpful when you are waiting for an important call. It lets you listen while a caller is leaving a message for you. **Message Center Keys** allow two people sharing the same phone to have their own message waiting key. Each person can see if he/she has new messages. **Conversation Record** saves and records your conversation into your mailbox with the touch of a button. **Park and Page** allows a caller to page you before leaving a message. You can pick up the call from any station.

Automatic Call Distribution (ACD) distributes calls evenly among member agents and provides initial and repetitive announcements that encourage callers to remain online. Callers can leave a message if they choose to receive a callback from an agent. Optional PC-based **Supervisor with Reports** can be used for agent scheduling.

Wireless/Cordless phones keep employees connected while away from their desk. Aspire provides a variety of mobility solutions to connect your whole campus or just your office.

System maintenance allows for online HTML-based programming access either on-site or over the Internet. Using browser software simplifies the process for changing names or speed dial settings. Special PC software is available for off-line programming and remote access by modem.

Call Logging saves information about incoming and outgoing calls. Logged calls can be redialed or saved to speed dial.

Aspire's **Automatic Route Selection (ARS)** system decides whether to place a call with a long distance carrier, over IP or, if allowed, a local trunk. You specify how you want your calls to be routed.

E911 Compatibility identifies the origination of an E911 call so emergency services can reach the specific extension location quickly.

Aspire Versatility Feature List

Usability

- Built-in Headset Jack
- Call Coverage Keys
- Caller ID Logging
- Contrast Control
- Last Number Redial List (10)
- One Touch Feature Operation
- Tilt Display

Flexibility

- Adjustable Height Telephone
- Flexible Numbering Plan
- Secure Station Relocation
- Universal Card Slots
- Virtual Extension Keys

Serviceability

- On-line Programming
- Remote Programming
- Self Diagnostics
- Single Pair Wiring

Versatility

- Analog Trunks and Stations
- Colored Face Mats
- Digital Trunks and Stations
- DID Trunks
- E&M Trunks
- IP Trunks and Stations
- i-Series Telephone Support
- T1/PRI Trunks
- TAPI Compatible

Scalability

- Application Processors
- IP Keysets
- IP Trunks
- Software Upgrades
- Universal Line/Station Card Slots

Manageability

- Account Codes, Forced & Verifiable
- Automatic Call Distribution
- Automatic Route Selection
- Built-in Mini Gatekeeper
- Conference Bridge
- IP Networking
- Networking
- Toll Restriction
- User Programming
- Walking Class of Service
- Web-based Programming

Adaptability

- In-switch Aspire
- Mail Cards
- Media Gateway
- Switching Hub
- In-server Card
- Wall, floor, 19” Rack Mountable Cabinet
- TDM and/or IP

The Telephones

✍ **Make a note. . .**
For your convenience. . . there is a Parts List located at the end of this guide. You should find this list helpful when selecting system equipment. More detailed tools are also available — ask your Account Representative for the specifics.

34-Button Super Display Telephone – P/Ns 0890049 & 0890050

2

The Super Display Telephone, which is supported with software 1.07 or higher, is the system's premier telephone instrument, featuring an interactive 9-line, 24-character display with 12 associated interactive keys. As the Super Display Telephone user processes calls, the interactive key functions change to provide intuitive access to the system's most sophisticated features. Every Super Display Telephone has a built-in speakerphone for full Handsfree operation. Handsfree Answerback and Intercom voice-announce capability is also standard.

The telephone's 24 programmable function keys can be customized by the user for one-button access to co-workers, features like Paging or Park or specific outside lines. The Dual LEDs in each programmable key help the user see which calls are for them and which features are active. Access to other commonly used features is simplified by 15 fixed feature keys.

In addition, the Super Display Telephone provides a built-in wall-mount bracket, as well as adjustable legs which allow each phone to be angled at a height which best suits the user.

Note that the three IP adapters (IP, ADA2, and PSA) do not work on the Super Display Telephones.

✍ **At a glance**

Super Display Telephone - Part Numbers 0890049 & 0890050

Function Keys:	✔	Accepts 110-Button DSS:	✔	Accepts 24-Button DLS:	✔
Handsfree (Speakerphone):	✔	Dual LEDs:	✔	ADA Adapter:	✔
ADA2 Adapter:	No	APA Adapter:	✔	APR Adapter:	✔
CTA Adapter:	✔	CTU Adapter:	✔	ILPA Adapter:	No
IP Adapter:	No	PSA Adapter:	No		

34-Button Display Telephone — P/Ns 0890045 & 0890046

The 34-Button Display Telephone has a 3-line, 24-character display with four interactive soft keys for intuitive feature access. In addition, the 34-Button Display Telephone has 24 user-programmable function keys (with Dual LEDs) for one-button access to co-workers, features and outside lines. The telephone also provides 10 user-programmable One-Touch (Personal Speed Dial) keys and 15 additional fixed feature keys.

The 34-Button Display Telephone has a built-in speakerphone and can accept optional adapters. You can also assign 110-Button DSS Consoles or connect 24-Button DLS Consoles to these phones. Like the Super Display, the 34-Button Display provides Handsfree Answerback, Intercom voice-announcements. In addition, the 34-Button Display provides a built-in wall-mount bracket, as well as adjustable legs which allow each phone to be angled at a height which best suits the user.

✍ **At a glance**

34-Button Display Telephone - Part Numbers 0890045 & 0890046

Function Keys:	✔	Accepts 110-Button DSS:	✔	Accepts 24-Button DLS:	✔
Handsfree (Speakerphone):	✔	Dual LEDs:	✔	ADA Adapter:	✔
ADA2 Adapter:	No	APA Adapter:	✔	APR Adapter:	✔
CTA Adapter:	✔	CTU Adapter:	✔	ILPA Adapter:	✔
IP Adapter	✔	PSA Adapter:	No		

22-Button Display Telephone — P/Ns 0890043 & 0890044

The 22-Button Display Telephone features a 3-line, 24-character display with 4 interactive soft keys for intuitive feature access, in addition to 12 function keys with Dual LEDs. The function keys are user-programmable and can provide 1-button access to co-workers, features and outside lines. The telephone additionally provides 10 user-programmable One-Touch (Personal Speed Dial) keys and 15 additional fixed feature keys.

The 22-Button Display Telephone has a built-in speakerphone, provides Handsfree Answerback, Intercom voice-announcements. In addition, the 22-Button Display provides a built-in wall-mount bracket, as well as adjustable legs which allow each phone to be angled at a height which best suits the user.

✍ **At a glance**

22-Button Display Telephone - Part Numbers 0890043 & 0890044

Function Keys:	✔	Accepts 110-Button DSS:	✔	Accepts 24-Button DLS:	✔
Handsfree (Speakerphone):	✔	Dual LEDs:	✔	ADA Adapter:	✔
ADA2 Adapter:	No	APA Adapter:	✔	APR Adapter:	✔
CTA Adapter:	✔	CTU Adapter:	✔	ILPA Adapter:	✔
IP Adapter:	✔	PSA Adapter:	No		

22-Button Standard Telephone — P/Ns 0890041 & 0890042

The 22-Button Telephone offers similar capabilities as the 22-Button Display Telephone, but excludes the alphanumeric display and soft keys.

✍ **At a glance**

22-Button Telephone - Part Numbers 0890041 & 0890042

Function Keys:	✔	Accepts 110-Button DSS:	✔	Accepts 24-Button DLS:	✔
Handsfree (Speakerphone):	✔	Dual LEDs:	✔	ADA Adapter:	✔
ADA2 Adapter:	No	APA Adapter:	✔	APR Adapter:	✔
CTA Adapter:	✔	CTU Adapter:	✔	ILPA Adapter:	✔
IP Adapter	✔	PSA Adapter:	No		

2-Button Telephone — P/Ns 0890047 & 0890048

The Digital 2-Button Telephone offers many keyset features and conveniences at an analog station set price. Handsfree Answer-back lets users answer Intercom calls without touching the phone. The 11 fixed feature keys provide quick access to many essential features, and the Message Waiting lamp always shows when there are unanswered messages.

The wall-mount bracket is not compatible with the standard telco (AT&T) wall plate.

✍ **At a glance**

2-Button Telephone - Part Numbers 0890047 & 0890048

Function Keys:	✔	Accepts 110-Button DSS:	✔	Accepts 24-Button DLS:	No
Handsfree (Speakerphone):	✔	Dual LEDs:	✔	ADA Adapter:	No
ADA2 Adapter:	No	APA Adapter:	No	APR Adapter:	No
CTA Adapter:	No	CTU Adapter:	No	ILPA Adapter:	No
IP Adapter:	No	PSA Adapter:	No		

Cordless Telephone — P/N 730088 & 730087

The Aspire System, with software 1.07 or higher, supports the Cordless Telephone. The DTR-4R-2 Cordless II (P/N 730088) is a 900 MHz spread-spectrum digital cordless telephone that provides mobility, flexibility and convenience for those who spend much of the workday away from their desk. Fully integrated with the telephone system, the DTR-4R-2 Cordless II offers many standard features such as Park, Do Not Disturb, Hotline, Voice Over and Voice Mail. Normally paired with a companion keyset for improved 1-button call coverage capabilities, the DTR-4R-2 Cordless II will also work as a stand-alone telephone.

Where users require greater range on the cordless phones and 3 or less cordless phones are being used at a specific site, the DTR-4R-2 Cordless II phone is recommended.

Complemented by 4 fully programmable function keys (with LEDs), the DTR-4R-2 Cordless II achieves a whole new level of convenience and mobility. An easy-to-read LCD display, volume controls, a rechargeable nickel-cadmium battery pack and a handy belt clip round out the elegant and affordable DTR-4R-2 Cordless II Phone. This phone provides 10 channels.

A second Cordless Phone is also available: the Cordless Lite II (P/N 730087). The Cordless Lite II offers the same features as the DTR-4R-2 Cordless II except that it uses a NiMH battery and has FM modulation (single channel) instead of the spread spectrum modulation.

Where more than 3 cordless phones are to be used at one specific site, the Cordless Lite II phone is recommended. This phone provides 30 channels.

Aspire Wireless (DECT) Phones — P/N 780004

The Aspire system, with software 2.16 or higher, provides the ability to use 2.4 GHz DECT (Digital Enhanced Cordless Telecommunication) phones. These phones provide you with the freedom and conveniences of a wireless phone, but in addition, you also have access to features provided by the Aspire system. These phones can not be used on the Aspire S system.

The Aspire Wireless (DECT) phone provides additional options as well.

- Alphanumeric Display with Back Light
- LED Indication for Incoming and Unanswered Calls
- Telephone Book with 80 Number Memory Capacity
- Vibration
- Auto Log-In (roaming between different systems)
- 14 Messages Stored
- Stack for 10 Caller ID
- Silent Mode (mute all sounds)
- Redial (last 10 numbers)
- Programming Pause
- Programming of 2 Different Setups (indoor and outdoor)
- Adjustable Volume
- Key Lock
- 9 Different Ring Tones and Adjustable Ring Volume
- Microphone Mute
- Headset Connection
- Automatic Off-Hook (B-Answer)
- R-Key for Transfer and Special Services

Keyset Adapters

Each Aspire keyset (except the 2-Button telephone) accepts the following optional adapters, though there are limitations with phone types (such as IP) and adapter compatibility. Each telephone can have up to two adapters installed except when an IP adapter is used. This adapter uses the full space on the keyset, preventing any other adapter installation. In addition, the IP adapter cannot be used on a Super Display telephone.

You can install one of these. . .

- **Call Recording Adapter (ADA) (P/N 0890055)**

Using the ADA Adapter provides a recording jack connection which provides a connection from a telephone to an external tape recorder or speaker.

- **Analog Interface with Ringing (APR) (P/N 0890056)**

The APR provides an analog interface for a keyset. The APR provides ringing which allows the connected device to be used for incoming and outgoing calls. It can use the same extension number as the keyset (B1 channel) or it can have its own extension number (B2 channel). The APR does not support reverse polarity, message waiting lamping, or Caller ID.

This adapter requires an AC Adapter (AC-2R), P/N 780135.

The APR for the B1 channel consumes no ports, however, the B2 channel consumes 1 port, ranging from 512 to 1 in descending order.

- **Analog Interface without Ringing (APA) (P/N 0890057)**

The APA provides an analog interface for a keyset. The connected device is used for outgoing calls only (for example, when using a modem). It does not provide ringing. When an analog device is in use, the keyset cannot be used (as the same port number is used for both devices). If both devices are picked up at the same time, the analog device takes priority. The APA does not support reverse polarity, message waiting lamping, or Caller ID.

- **RS-232C Adapter (CTA) (P/N 0890058)**

Provides a serial interface (RS-232C) connector. This can be used for SMDR or TAPI (1.4) or system reporting.

- **Speakerphone Adapter (HF-R) (P/N 0890062 & 0890063)**

Offers 22-Button, 34-Button, and Super Display keysets high quality speakerphone capability.

This adapter requires an AC Adapter (AC-2R), P/N 780135.

- **USB Adapter (CTU) (P/N 0890059)**

Provides a USB connector. This can be used for either SMDR, TAPI (1.4) or system reporting.

This adapter requires an AC Adapter (AC-2R), P/N 780135.

- **24-Button DLS Console (P/N 0890053 & 0890054)**

Refer to the “110-Button DSS Console and 24-Button DLS Console” on **24-Button DLS Console — P/Ns 0890053 & 0890054** (page 12) for more details.

- **IP Adapter (P/N 0890060)**

Provides the ability to communicate through a LAN which is connected to a VOIPU or 8SHUBU PCB. The VOIPU or 8SHUBU PCB is required in order to communicate with non-VoIP Aspire phones, as well as to place or receive outside calls. This adapter **cannot** be used on a Super Display telephone.

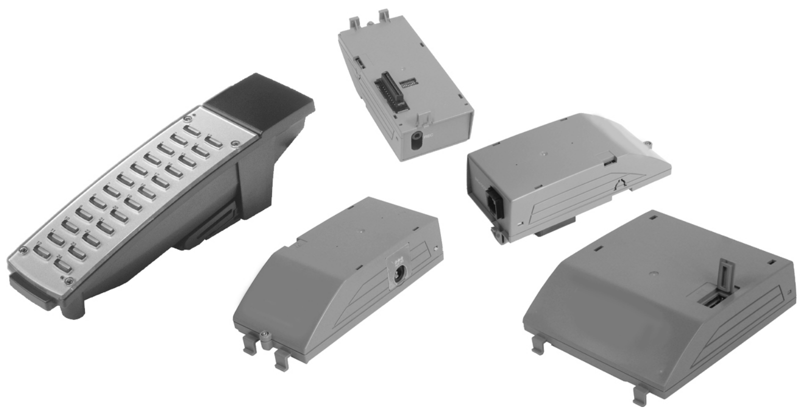
This adapter requires an AC Adapter (AC-2R), P/N 780135.

- **IPhone Power Failure Adapter (PSA) (P/N 0890067)**

Provides power failure capacity for the IPhone IP keysets. This allows a user to make or receive an outside call using the central office if an IP keyset is unable to make or receive a call using the LAN.

- **IP Call Recording Adapter (ADA2) (P/N 0890066)**

Using the ADA2 Adapter provides a recording jack connection which provides a connection from an Aspire IPhone to an external tape recorder or speaker.



The following chart indicates if there are restrictions when combining certain adapters. Select the adapter in the column and then select the adapter in the row to see if there are any restrictions. For example, using an APA and APR adapter refers you to restriction 3 (only one voice path provided - adapters can not be used together)

Adapter Compatibility Restrictions

	24DLS	IP	CTU	CTA	APR	APA	ADA	ADA2	HF-R	PSA
ADA	√	1	√	√	√	√	2	5	√	5
ADA2	5	5	5	5	5	5	5	5	5	5
APA	√	1	√	√	3	2	√	5	√	5
APR	√	1	6	√	2	3	√	5	6	5
CTA	√	1	4	2	√	√	√	5	√	5
CTU	√	1	2	4	6	√	√	5	6	5
IP	1	2	1	1	1	1	1	5	√	5
HF-R	√	1	6	√	6	√	√	5	2	5
PSA	5	5	5	5	5	5	5	5	5	5
24DLS	2	1	√	√	√	√	√	5	√	5

Adapter Compatibility Restrictions

1 = The IP Adapter takes the full space provided for adapters on the keysets. Therefore, if an IP adapter is installed, no other adapters can be used.

2 = Only one adapter of the same type can be used on a keyset.

3 = As there is only one voice path provided for adapters, the APR and APA adapters can not both be used on the same keyset.

4 = Due to protocol collision, the CTU and CTA adapters can not both be used on the same keyset.

5 = The ADA2 and PSA Adapters can only be installed on the Aspire IPhone, which only has one adapter connection. Therefore, if either the ADA2 or PSA is installed, no other adapters can be used.

6 = As this adapter requires the AC power adapter, it can not be installed on a phone with an APR or Speakerphone adapter, which also requires power. The placement of the AC power adapter plug will not allow the unit installed on the left side of the phone to receive power.

When installing the adapters, ***the keyset should first be unplugged from the system***. Also note that the adapters may have an AC/DC power jack. Power is not required for all the adapters. You should refer to the information for the specific adapter to determine whether a power source is needed.

Telephones with any of these adapters installed cannot be wall-mounted. The bracket will not accommodate the adapter(s).

Optional Equipment

2PGDAD Module — P/N 0891027

The 2PGDAD module provides two circuits which allow connection to external terminals such as:

- Door Box
  (Aspire S: 4 max. per system, Aspire: 8 max. per system)
- External Speaker with Amplifier
  (Aspire S: 4 maximum with 2PGDAD modules, Aspire: 8 maximum with 2PGDAD modules, 1 on the NTCPU)
- External Music Source (external MOH)
  (Aspire S: 8 maximum per system, Aspire: 96 maximum per system)
- External Recording System
  (Aspire S: 8 maximum per system, Aspire: 96 maximum per system)
- External Ringing

The Aspire S supports a total of 10 2PGDAD modules on the system which can be used within the feature maximums indicated above. The Aspire supports a total of 56 2PGDAD modules on the system which can be used within the feature maximums indicated above.

The 2PGDAD module also provides multi-purpose controls. These control relays can be used for controlling the external amplifier, external music source and door lock control with the use of a Door Box. The Aspire S system allows for up 8 general purpose relays with the 2PGDAD modules. The Aspire system allows for up 8 general purpose relays with the 2PGDAD modules (4 relays on each 2PGDAD) and 1 on the NTCPU for a maximum of 9.

The 2PGDAD module connects to any available digital extension port. The terminal connections made within the PGDAD module and the jumper settings determine what features are used for each circuit.

110-Button DSS Console — P/Ns 0890051 & 0890052
24-Button DLS Console — P/Ns 0890053 & 0890054

The DSS (Direct Station Selection) Console gives a keyset user a Busy Lamp Field (BLF) and one-button access to extensions, trunks and system features. The 110-Button DSS Console provides an additional 100 programmable keys, while the 24-Button DLS Console provides 24 programmable keys. The 110-Button DSS also has 10 fixed feature keys for Paging, calling Door Boxes, activating Night Service and enabling DSS Console Alternate Answer. There are also two keys that allow “shifting” between the first and second set of 100 extensions.

Keep the following in mind when installing DSS Consoles:

- A 110-Button DSS Console requires a separate digital station port.
- A 24-Button DLS Console does **not** require a separate station port.

- The system allows for a maximum of 4 (Aspire S) or 32 (Aspire) 110-button DSS Consoles. One extension can have a maximum of 4 110-button DSS Consoles. As the 24-button DLS Console is connected to the bottom of the phone, an extension can only have one 24-button DLS Console installed, but each extension in the system can have a 24-button Console (maximum of 24 on Aspire S or 256 on Aspire). An extension can have a 24-button DLS Console and 4 110-button DSS consoles installed/assigned.

- A 24-button DLS Console cannot be installed on an IPhone.

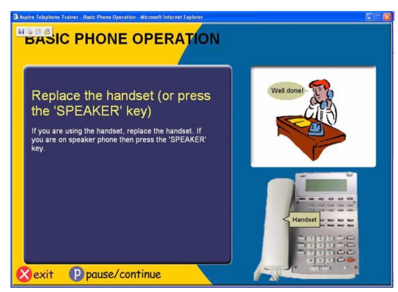
- By default, the 24-Button DLS Console has no keys defined. These keys can be programmed as line keys, extension DSS keys, or programmable function keys using Program 15-07. To program the keys, use the extension number to which the DLS is installed and, regardless of the type of keyset connected, ***start programming the DLS keys at key number 25***. Service codes 851 and 852 can also be used to program these keys if allowed by an extension's Class of Service.

- By default, the 110-Button DSS Console has extension DSS keys defined. These keys can be programmed as line keys, extension DSS keys, or programmable function keys using Program 30-03-01. Service codes 851 and 852 can also be used to program these keys if allowed by an extension's Class of Service.

For additional information, refer to *Direct Station Selection (DSS) Console* in the Software Manual.

Aspire Telephone Trainer

The NEC Aspire Telephone Trainer is a revolutionary end-user training product designed to ease the burden of reaching and training telephone users as part of a new telephone system installation. The Aspire Telephone Trainer is also a valuable tool for the customer to train new staff (or to re-train existing staff) during the lifetime of the PBX.



If configured and used correctly, the Aspire Telephone Trainer will improve your customer service and customer satisfaction.

Below are just a few specific benefits of the Aspire Telephone Trainer over traditional telephone training services:

- Always available – Aspire Telephone Trainer is available to the customer 24 hours a day, seven days a week. Employees can train in the office or at home.
- No PBX required – Aspire Telephone Trainer training is "off-line" which means you can train before a new telephone system goes live and ensure training is non-disruptive to other telephone users.
- Fast one-on-one training – It takes typically less than 20 minutes to thoroughly train an individual on a personalized basis.
- Ideal for new employees – The customer can implement telephone training as part of the orientation program for new employees without calling in a specialized trainer.
- Control – The customer can decide which User Groups train on which features.

Aspire Telephone Trainer comprises the following key components:

- Enabler – The Aspire Telephone Trainer Enabler is a device that connects to the customer's PC to drive the Aspire digital telephone. Each Enabler allows different users to log in and receive training at the same time.
- AC Adapter – The adapter plugs into an external power source and into the Enabler box providing power for the keyset.

- USB Cable – The USB cable is used to connect the Enabler to the PC.
- Telephone Line Cord – The line cord connects the telephone to the Enabler.
- CD – The CD provides the Aspire Telephone Trainer software which is loaded into the PC as well as the Acrobat files of the Telephone Trainer documentation.
- Installation and Administration Guide – Describes the installation, setup and operation of the Aspire Telephone Trainer.

Audio Emcee Kit — P/N 750316

The Audio Emcee message on hold system provides the ability to use a CD-based on-hold messaging system connected to the Aspire through the music on hold (MOH) port. When business calls are placed on hold, callers will hear advertisements that can help enhance the company image, cross sell products and services and reduce caller hangs-ups by keeping callers entertained and informed. Callers will hear programmed selections and this cycle will repeat until the call is answered.

The kit includes:

- MOH Unit
- Infrared Remote
- RCA Connecting Cable
- Starter CD with 24 Message/Music Tracks
  (8 music tracks, 8 courtesy tracks, 8 holiday tracks)
- Certificate Good for 8 Customized Messages
- Operating Guide
- 1 Year License

Door Box — P/N 92245

The Door Box is a self-contained Intercom unit typically used to monitor an entrance door. A visitor at the door can press the Door Box call button (like a door bell). The Door Box then sends chime tones to all extensions programmed to receive chimes. The Door Box is weather-tight, but where possible, should have some coverage from the weather. A 2PGDAD Module is required for this feature.

Each 2PGDAD module audio output can optionally support two analog Door Boxes. In addition, you can connect each circuit's control relay to an electric door strike. This allows an extension user to remotely activate the door strike while talking to a visitor at the Door Box. The control relays are normally open. The NTCPU also provides 1 relay. This relay is defined as relay '0' in programming. The relays on the 2PGDAD modules are numbered 1-8. The system can have up to four (Aspire S) or eight (Aspire) Door Boxes.

Headsets

Headsets are perfect for users who spend a lot of time on the phone. They enable users to become more productive by freeing their hands while talking on the phone and also prevents neck strain from trying to balance the handset between their ear and shoulder. The following modular headsets are compatible with the Aspire system.

- Polaris™ SupraPlus® Monaural, Noise Cancelling (P/N 750643)
- Polaris™ SupraPlus® Binaural, Noise Cancelling (P/N 750645)
- Polaris™ SupraPlus Monaural - Voice Tube (P/N 750644)
- Polaris™ Encore Monaural - Voice Tube (P/N 750634)
- Polaris™ Encore® Binaural, Noise Cancelling (P/N 750635)
- Polaris Tristar - Voice Tube (P/N 750630)
- Polaris Mirage - Voice Tube (P/N 750631)
- CT-11 Cordless Headset Telephone (P/N 730089)
- Dterm Headset Cordless II (P/N 730091)

CT-11 Cordless Headset Telephone (P/N 730089)

The CT-11 is a 2.4GHz cordless headset which connects to an analog port or an analog telephone line as a stand-alone unit or to an analog port adapter (APR, P/N 0890056). When the APR is set up as the same extension of the telephone, you can use the headset to answer and make calls using the cordless headset. The CT-11 offers Caller ID, but only if it is connected to an analog port on an analog station card. The CT-11 will not receive Caller ID if it is connected to an APA or APR adapter (these adapters do not output Caller ID).

The number of units which can be used on the system is greatly affected by the environment. The closer or smaller the area, the smaller the number of units which can be used. It is recommended to start with 3 or less. If there are no conflicts between the telephones, you can try adding additional units (up to 5 would be the recommended maximum).

When using wireless LAN, keep in mind that although there should not be a problem with interference from WLAN's, 802.11b and 802.11g both share the same frequency as the CT-11 telephone. In theory, the CT-11 is a narrow band high power device where as the 802.11b and 802.11g are both wide band low power technologies. Therefore, the higher power CT-11 could disrupt the low power device and slow the data network. There are, however, many exceptions to this (for example, if the WLAN uses highly directional antennas, higher power relays between buildings, etc.). The CT-11 can not lock down channels, unlike the 802.11b and 802.11g.

The CT-11 features include:

- 2.4 GHz Cordless Headset Phone
- Range of Up to 150’
- 6 Hours of Talk Time, 80 Hours Standby Time
- Audible Low Battery Indicator
- Single Line Operation
- Ultra-Compact Remote Unit with Belt Clip
- Variable Range Volume Control
- 10 Speed Dial Numbers
- Page/Find Feature
- Redial/Flash
- Mute with Audible Reminder
- Talk/Charge/Power Indicator Lights
- Built-in Headset Stand

Dterm Headset Cordless II (P/N 730091)

The Dterm Headset Cordless II is a cordless headset telephone. It is basically a single-line telephone, but it has no dial, so it does not provide all of the functions of a single-line telephone. As such, it generally would not be connected directly to a telephone line or an analog station port. It is specifically designed to interface with an Aspire 22/34-button keyset equipped with an APA or APR Adapter.

The number of units which can be used on the system can range from 15-50. The maximum number is greatly affected by the environment. The closer or smaller the area, the smaller the number of units which can be used. The Dterm Headset Cordless II uses frequency hopping to automatically change channels if interference is present. If the unit detects wideband interference (multiple users in a small area), the range will be decreased until an adequate link is formed. The present of other 900 MHz devices will cause interference and may need to be removed.

The Dterm Headset Cordless II features include:

- 8 hours of talk time
- 902-928MHz DECT Wireless Technology
- Up to 300’ roaming distance from an office phone
- Can be converted from “over-the-head’ style to “over-the-ear’ style
- The base station is equipped with:
  - 4 audio adjustments
  - 3 status LEDs (Power, Talk, and Charging)
  - Handset Lifter switch (not used on the NEC version)
  - Charging cradle for the headset
  - AC adapter
  - 1 connecting cord
  - 1 DTR-1C-2 swtich cord
- The headset is equipped with:
  - Noise canceling microphone
  - Talk button (answer or disconnect a call)
  - Talk indicator LED
  - Listening volume control/mute switch
  - 2 styles of headbands with 2 types of foam ear rests
  - 3 different size ear loops

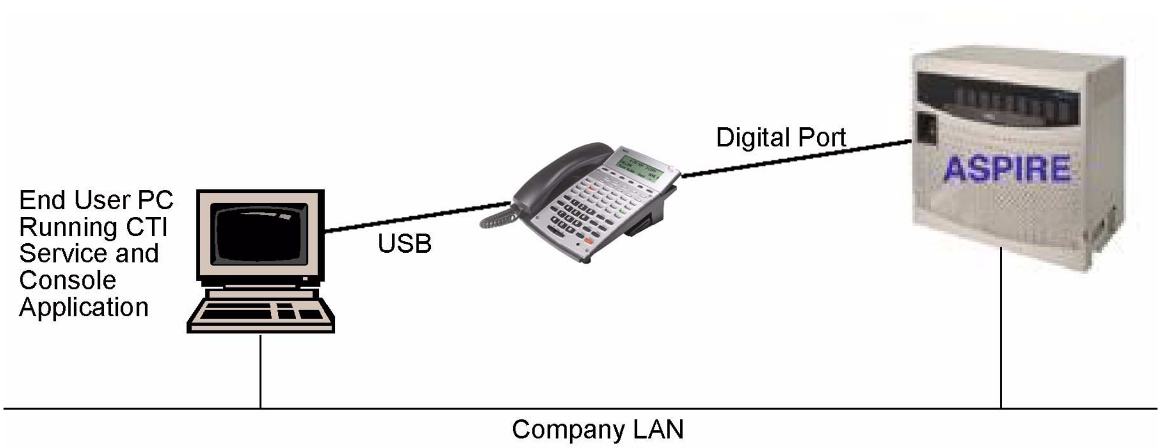
PC Attendant

The PC Attendant Console is a software product that provides access to the most common functions required by an operator or receptionist. By using this application in conjunction with an Aspire S, M, L, or XL system, the attendant can easily manage their call handling tasks without having to switch their attention between the telephone and the PC. A company directory, recording capability, and PC-to-PC messaging, provide additional features to further enhance the operation. (The attendant telephone requires a CTU Adapter (P/N 0890059) installed with connection directly to the end-user PC for the Call Recording and Personal Greeting voice functions.)

The PC Attendant application uses a CTI (Computer Telephony Integration) service that is provided with the product in order to integrate with the Aspire telephone systems. Through the CTI service, an ethernet connection over the company LAN, and an Aspire keyset for audio, the PC Attendant application is able to monitor all extensions on the phone system and control the actions of the attendant's phone, including placing calls.

The CTI service on the PC communicates with the Aspire system through the CTI port on the telephone system. An administration utility (Telephony Administrator) is provided which allows the system administrator to configure the global settings for the console application.

The following diagram shows the system configuration for a single console installation.



The PC Attendant application can also be installed on multiple PCs for installations that need to support more than one attendant position (each attendant PC would require a licensed version of the PC Attendant installed). ***Up to 8 PC Attendant positions can be installed.***

The PC Attendant application also includes a supporting application, call Quick Message. By installing the Quick Message client on individual PCs, the attendant is able to quickly send short messages to other employees, who can respond with a single keystroke.

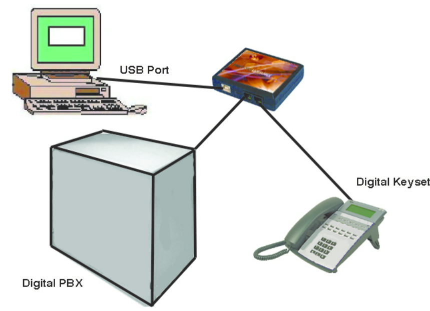
The host PC to be used as the CTI Server requires Windows XP, Windows 2000, or Windows 2003 and an interface to the Aspire system through the 3rd-party CTI link to monitor and control the telephone activity (*Windows Small Business Server 2003 is not currently supported by the Aspire CTI/TAPI server*). When installing the PC Attendant Console on multiple PCs for more than one attendant position, the PC requires Windows XP, Windows 2000 or Windows 2003. (Note that if the client PC is to be used for other TAPI 2 applications (ex: Microsoft Outlook dialing), you would need to use Windows 2000 or 2003 - Windows XP will not support all TAPI 2 applications.

SLT Adapter — P/N 0891026

The SLT Adapter converts a digital port from an ESIU PCB into an analog port which can be used for connecting on-premise 2500 type single line devices (i.e., telephones, fax machines, modems, etc.). Caller ID is supported by these adapters. The SLT Adapter provides the ring generator circuit used by the analog device. The unit provides constant current which is fixed at 47 mA. Each SLT Adapter requires its own digital port.

Refer to *Single Line Telephones* in the Software Manual for more details.

VSR Adapter — P/N 0891026



The Dterm® Voice Security Recorder (P/N 780275) is a USB device that taps across the digital extension pair of the NEC telephone system allowing digital recording of the keyset user's conversation. The file created is saved either to the local PC or to a network location, depending on the application's setup. This adapter is for use with digital keysets. It cannot be used with analog, VoIP or i-Series phones.

CAUTION
The use of monitoring, recording, or listening devices to eavesdrop, monitor, retrieve, or record telephone conversation or other sound activities, whether or not contemporaneous with transmission, may be illegal in certain circumstances under federal or state laws. Legal advice should be sought prior to implementing any practice that monitors or records any telephone conversation. Some federal and state laws require some form of notification to all parties to a telephone conversation, such as using a beep tone or other notification methods or requiring the consent of all parties to the telephone conversation, prior to monitoring or recording the telephone conversation. Some of these laws incorporate strict penalties.

PC Compatibility

The Dterm® Voice Security Recorder application supports Microsoft operating systems which support USB devices such as Windows 98SE, Windows ME, Windows 2000, and Windows XP. Note that Windows 95 and below, Windows NT and Macintosh operating systems are not supported.

Aspire S Common Equipment

Aspire 6 Slot KSU w/ CPU and PS — P/N 0890005

The KSU is the system's control center. It houses the Power Supply, CPU, has six PCB slots and provides for connection to trunks and extensions. The slots are universal. They can be used for any combination of Common, Trunk or Station PCBs. The CPU controls all the functions and operations of the Aspire system using the system software loaded into the CPU memory. The KSU can be floor, wall or rack mounted. The CPU, which is pre-installed, controls all the functions and operations of the Aspire system using the system software loaded into the CPU memory. One 32-bit CPU is installed in the system cabinet.

2

The CPU provides the following:

- Accommodates up to 34 ports (8 trunks x 26 extensions)
- 8 digital station connections

  A 2PGDAD module cannot be connected to port 1 or port 2.
- 2 analog station connections (no Message Wait lamping)
- 4 diagnostic LEDs which indicate the status of various system functions

  During normal operation, the "LD2" LED will be flashing. The remaining LEDs can flash on or off depending on the current system operation.
- Time Switch (383 ch)
- Digital Phase Locked Loop (DPLL): digital phase synchronization loop
- SFLM Generation
- DSP (Digital Signal Processor: provides C-Channel control
  - Tone Generation
  - DTMF Tone Sender/Receiver
  - System Tone Sender
  - MFC Tone Sender
  - MF Signal Sender (Sends caller information to CO for E911)
  - Call Progress Tone Detector
  - C-Channel Control
  - Time Switch control
  - HDLC (High-Level Data Link Control) Packet Processing
  - Conference; 32 Channels
  - Caller ID Receiver/Generation; 16 Channels
- inDepth Support (requires software 2.65+) - ***Does not include ACD support***
- TAPI 1.x/TAPI 2 Support
- A load button which is used for initial system startup or when upgrading system software
- One Serial Port
- One Compact Flash Card Slot
- One Audio Input Terminal (external MOH/BGM source)
- General Purpose Control Terminal
- Hold Tone Transmit
- IP
- Real Time Clock (tolerance 30 seconds/month)
- Internal MOH Generation
- One Connector for PAL EPROM
- One lithium battery (Sony CR2032 or equivalent) which provides battery back-up of system data and RAM memory for approximately 30 months

Floor/Desk Stand Mounting Brackets — P/N 0891303

The Floor/Desk Stand Mounting Brackets are used to secure the KSU in an upright position to either a floor or desk area.

19” Rack Mount Bracket — P/N 0891300

The 19” Rack Mount Bracket is required to install the KSU onto a 19” rack mounting system.

DSP Resource Daughter Board (DSPDB) — P/N 0891003

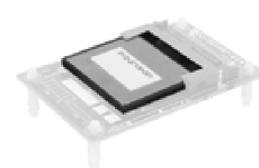
The DSPDB provides the option for the VRS (Voice Response System) feature. This daughter board is mounted on the CPU and provides:

- 8 VRS Circuits with a VRS Flash Card Installed (replays up to 8 circuits simultaneously; records up to 8 circuits simultaneously)
- Compact Flash Slot for VRS Feature

Note: The DSPDB does not provide any additional resources as with the Aspire M/L/XL.

Flash VRS CF (DSPDB Compact Flash Card) (DSPDB) — P/N 0891040

The Flash Card adds Voice Announce/Automated Attendant features and is installed on the DSP Resource daughter board.



LAN Connection (ENTU) PCB — P/N 0891053

An ENTU PCB is required when a VoIP PCB is installed. If the ENTU is not installed, the system will not start up.

The ENTU PCB provides a LAN connector which is compatible with 100Base-TX and 10Base-T. This PCB is compatible in LAN applications using 10Mbps and 100Mbps. All ports will automatically identify and switch 100Base-TX, 10Base-T and Full/Half Duplex. This PCB is required for connecting with PCPro/WebPro via a LAN connection.

The VoIP PCB, which is required in order for IP telephones to communicate with non-VoIP Aspire phones, as well as to place or receive outside calls, must be connected to an external switching hub.

The PCB plugs into the CN6 and CN7 connectors on the CPU, with a maximum of 1 per system. Each PCB provides an RJ61 port connector. This is used to connect to a LAN terminal or external switching hub. Depending on the type of LAN terminal, the PCB may not be able to detect the difference between straight cable and cross-cable automatically. If auto-crossover is not functioning, use straight cable for that terminal connection.

If PoE (power over ethernet) is to be used to eliminate the separate power adapters, a separate power source is required. It is recommended that you use a power switch and/or power hub which is IEEE 802.3AF compliant. For example, the PowerDsine 6xxx series of products is unique in the fact that in addition to offering IEEE 802.3AF support, it also provides for the NEC proprietary detection and, therefore, the ILPA (In Line Power Adapter) is not required. For systems which require layer 2 switching capability and PoE, the NEC BlueFire 200/24 switch is recommended. This unit provides layer 2 switch capability in addition to being able to supply ethernet power to 24 NEC IP terminals. For this unit, power feeding is through the signal pair (1/2, 3/6) or spare pair (4/5, 7/8).

Aspire S Trunk PCBs

4 CO Loop Start Trunk Card (4COIU-S) — P/N 0891046

The Analog Trunk (4COIU) PCB provides:

- 4 analog loop start line/trunk circuits - no ground start is provided
- 4 trunk status LEDs
- 4 Caller ID Circuits
- 1 Power Failure Transfer Circuits
- 1 PCB status LED
- 1 PABX Grounding Wire

The CN5 and CN7 connectors each provide connection to 2 analog trunk ports, ***which are polarity sensitive (tip to tip, ring to ring)***. The power failure circuit, however, is not polarity sensitive. A maximum of 2 4COIUs per system is allowed.

2 DID/OPX (2DIOPU-S) Card — P/N 0891047

The 2DIOPU PCB supports the analog DID and single line telephone interface functions (such as Off-Premise Extension). The function type is assigned in programming for each port.

The DIOPU PCB provides:

- 2 DID/OPX trunk circuits
- 2 DID/OPX trunk status LEDs
- 1 PCB status LED
- 1 PBXG Grounding Wire

The CN201 connector provides connection to 2 analog DID trunk ports, ***which are polarity sensitive (tip to tip, ring to ring)***. The OPX circuits, however, are not polarity sensitive. The DIOPU requires one universal slot, with 4 maximum PCBs per system. The system will assign either trunk or extension ports to the PCB based on the system programming (10-03-01). The PCB provides track resistance of 1500 ohms (PB loop) and 3000 ohms (DP loop) which includes the DC resistance of the terminal. This PCB requires system software 2.21 or higher.

Aspire S Station PCBs

8 Aspire Digital Card (8ESIU-S) — P/N 0891051

The 8ESIU PCB provides:

- 8 digital extension circuits (used for digital telephones, DSS consoles, 1SLTAD adapters, 2PGDAD adapters)
- 1 extension status LED (indicates status for 4 extensions)
- 1 PCB status LED

The CN3 and CN4 connectors each provide connection to 4 digital station ports. The ESIU requires one universal slot, with a maximum of 2 PCB's per system. *Note that this PCB can not be installed in slot 7 or 8.*

The 8ESIU consumes 8 ports ranging between ports 001-26. There must be enough available ports in order for the Aspire system to recognize an 8ESIU PCB.

4 Analog Station Card (4SLIU-S) — P/N 0891048

The 4SLIU PCB provides:

- 4 analog extension ports (used for on-premise analog telephones, fax machines, and analog modems)
- 4 extension status LEDs
- 1 PCB status LED
- Connector for 4SLIDB Daughter Board
- Ring Generator
- Message Wait Lamping Ability

Note: When connecting a fax machine or analog modem, make sure to set Program 15-03-03 to '1' (special terminal) to avoid communication problems.

The CN3 connector provides connection to 4 analog station ports and are not polarity sensitive. The 4SLIU is installed in a universal slot with a maximum number of 4 PCBs per system. This number of PCBs installed will reduce the number of 4SLIDB PCBs which can be installed. If 3 4SLIU PCBs are installed, only 1 4SLIDB can be installed. If 4SLIU PCBs are installed, the 4SLIDB can not be used. There must be enough available ports in order for the Aspire system to recognize an 4SLIU PCB.

4 Analog Station Expansion Daughter Board (4SLIDB-S) — P/N 0891049

The 4SLIDB daughter board provides:

- 4 analog extension ports (used for on-premise analog telephones, fax machines, and analog modems)
- Connector for 4SLIU PCB
- Ring Generator
- Message Wait Lamping Ability

Note: When connecting a fax machine or analog modem, make sure to set Program 15-03-03 to '1' (special terminal) to avoid communication problems.

The CN3 connector provides connection to 4 analog station ports and are not polarity sensitive. The 4SLIDB is installed on the 4SLIU PCB. Up to 2 PCBs can be installed in the system maximum, but this number may be reduced depending on how many 4SLIU PCBs are installed. If 3 4SLIU PCBs are installed, only 1 4SLIDB can be installed. If 4SLIU PCBs are installed, the 4SLIDB can not be used.

Voice Mail

4 Port IntraMail Compact Flash Card — P/N 0892180

The IntraMail is a plug-in “in-skin” full-featured, DSP-based integrated Voice Mail with Automated Attendant for the Aspire S (with software 2.50 or higher). This card is installed on the DSP Resource daughter board.

The 4 port compact flash card provides:

- 4 Voice Mail ports, 8 hours of message storage, and up to 160 mailboxes.
  It requires a DSPDBU Daughter Board P/N 0891003.

The IntraMail Automated Attendant answers incoming calls and routes them quickly and efficiently. Integrated Voice Mail features include Conversation Record, Answering Machine Emulation, and Caller ID with Return Call. Interactive Soft Keys guide the display telephone user through the extensive IntraMail feature set. This feature is available in PCPro software 2.30 or higher.

8 Port IntraMail Compact Flash Card — P/N 0892182

The IntraMail is a plug-in “in-skin” full-featured, DSP-based integrated Voice Mail with Automated Attendant for the Aspire S (with software 2.50 or higher). This card is installed on the DSP Resource daughter board.

The 8 port compact flash card provides:

- 8 Voice Mail ports, 16 hours of message storage, and up to 160 mailboxes.
  It requires a DSPDBU Daughter Board P/N 0891003.

The IntraMail Automated Attendant answers incoming calls and routes them quickly and efficiently. Integrated Voice Mail features include Conversation Record, Answering Machine Emulation, and Caller ID with Return Call. Interactive Soft Keys guide the display telephone user through the extensive IntraMail feature set. This feature is available in PCPro software 2.30 or higher.

IntraMail Specifications		
	0892180	**0892182**
Ports:	4	8
Station Mailboxes:	128	
Routing Mailboxes:	16	
Master Mailboxes:	16 (Only 8 of the 16 Master mailboxes are accessible in Aspire S.)	
Total Mailboxes:	160	
Storage Hours:	8 Hours	16 Hours
Answer Tables:	8	
Dial Action Tables:	16	
Programming Interface:	Aspire S telephone programming or Aspire PCPro software 2.30+.	
Remote Programming:	Access Via HTML-based Aspire WebPro or using customer-provided modems with Aspire PCPro.	
Voice Storage Media:	Flash Card (on IntraMail PCB)	
Languages:	1 (English Mnemonic)	

IP Interface Cards

4 CH VoIP Media Gateway (4VOIPU-S) — P/N 0891054

An ENTU PCB is required when a VoIP PCB is installed. If the ENTU is not installed, the system will not start up.

The 4VOIPU PCB is used for converting the RTP (Real Time Transfer Protocol) packets via the IP network and PCM highway. The IP telephones are connected directly to the IP bus. When IP phones need to be connected to a conventional PCM-based digital circuit, this PCB converts the IP packet signal into a PCM signal format and connects to the PCM time division switch.

The VOIPU PCB is required in order for IP telephones to communicate with non-VoIP Aspire phones, as well as to place or receive outside calls.

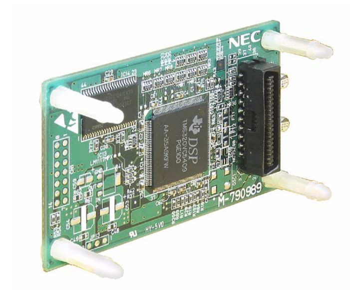
The 4VOIPU PCB provides:
- 4VOIPU PCB provides up to 4 channels
- Connector for the 4VOIPDB daughter board (providing an additional 4 channels)
- 1 PCB status LED
- DB status LED
- RIP session status LED
- PCMCIA slot LED
- Reset switch
- Load switch
- Boot jumper
- 1 run/block switch

A maximum of 3 PCBs per system are allowed. This provides 12 channels per system with the 4VOIPU (with the 4VOIPDB, 24 channels are available).

The separate software hub should be a 100Base/full duplex hub. To avoid network problems and to ensure good voice quality, do not use a Repeater Hub/10Base.

The Aspire VoIP supports H.323, H.325, and H.245 trunks and compressions of G.711, G.723.1, and G.729.

4 CH VoIP Media Gateway Expansion Daughter Board (4VOIPDB-S) — P/N 0891055



The VOIPDB daughter board provides:

- 4 channels
- Connector for the 4VOIPU PCB (combination provides a maximum of 8 channels per slot)

The VOIPDB is installed on the VOIPU PCB with a maximum of 3 daughter boards per system. This provides a maximum of 24 channels when combining the 4VOIPU and 4VOIPDB.

When installing a 4VOIPU with a 4VOIPDB PCB, the system uses a consecutive block of 8 trunk ports if there are 8 ports available.When installing a VoIP PCB, the system automatically assigns trunk ports to match the card's port capacity. For example, the 4VOIPU with a 4VOIPDB would take 8 trunk ports. Extension ports are not reserved until an IP phone is connected to the system. When the first IP phone is plugged in, the system takes the next four consecutive extension ports available and automatically assigns them as IP ports. The next three IP phones installed will use this group of ports. When the fifth IP phone is connected, the next 4 consecutive extension ports available will be assigned as IP ports.

If the number of trunk ports reserved by the system is a concern (as it could be with the Aspire S system), install the trunk cards first, then install the VOIPU PCB. This will allow the trunks to be assigned to the COIU, DIOPU, etc. first. If there are not enough trunk ports available for the VoIP PCB, the system will still recognize the card and allow it to be used for IP phones.

If the PCB is not going to be used for trunks, the logical trunk ports can be set to '0' in **Program 10-03-01 : PCB Setup**, but the physical trunk ports are still assigned to the PCB and cannot be used for any other PCB unless the PCB is deleted from the slot in **Program 90-05 : Slot Control**.

The Aspire VoIP supports H.323, H.325, and H.245 trunks and compressions of G.711, G.723.1, and G.729.

IP Station Equipment

34-Button Aspire IPhone - BK — P/N 0890065

This keyset provides a network connector which allows it to be used with the VoIP feature.

The 34-Button Aspire IPhone has a 3-line, 24-character display with four interactive soft keys for intuitive feature access. In addition, it has 24 user-programmable function keys (with Dual LEDs) for one-button access to co-workers, features and outside lines. The telephone also provides 10 user-programmable One-Touch (Personal Speed Dial) keys and 15 additional fixed feature keys.

The 34-Button Aspire IPhone has a built-in speakerphone and can accept optional IP adapters (PSA, ADA2). You can also assign 110-Button DSS Consoles to these phones, but they must be on site (not networked). It provides Handsfree Answerback, Intercom voice-announcements. In addition, the telephone provides a built-in wall-mount bracket, as well as adjustable legs which allow each phone to be angled at a height which best suits the user.

2

✍ **At a glance**

34-Button Aspire IP Telephone - Part Number 0890065

Function Keys:	✔	Accepts 110-Button DSS:	✔	Accepts 24-Button DLS:	No
Handsfree (Speakerphone):	✔	Dual LEDs:	✔	ADA Adapter:	No
ADA2 Adapter:	✔	APA Adapter:	No	APR Adapter:	No
CTA Adapter:	No	CTU Adapter:	No	HF-R Adapter	No
ILPA Adapter	✔	:IP Adapter	No	PSA Adapter	✔

4-Button Aspire PoE IP Phone - BK — P/N 0890072

This 4-button IP keyset is a 802.3af PoE-compliant phone which provides a network connector, allowing it to be used with the VoIP feature.

The 34-Button Aspire IP phone has a 3-line, 24-character display with four interactive soft keys for intuitive feature access. In addition, it has 4 user-programmable function keys (with Dual LEDs) for one-button access to co-workers, features and outside lines. The telephone also provides an 13 additional fixed feature keys.

The 4-Button Aspire IP phone has a built-in speakerphone and provides Handsfree Answerback and Intercom voice-announcements. In addition, the telephone provides a built-in wall-mount bracket.

This phone does **not** provide:

- Data connector for the PC

  Only one ethernet port provided which is used to connect the phone to the system.
- Option Connector
- Headset Connector
- One-Touch Keys

  The One-Touch bins can be accessed, however, through the Soft Keys.
- CONF Key

  Conference can be set up using the Soft Keys or you can define a CONF key on one of the Programmable Function Keys.
- MSG Key

  The Message function can be accessed using the dial pad or through a Programmable Function Key (depending on the action being taken).

✍ **At a glance**

4-Button Display Telephone - Part Number 0890072

Function Keys:	✔	Accepts 110-Button DSS:	✔	Accepts 24-Button DLS:	No
Handsfree (Speakerphone):	✔	Dual LEDs:	✔	ADA Adapter:	No
ADA2 Adapter:	No	APA Adapter:	No	APR Adapter:	No
CTA Adapter:	No	CTU Adapter:	No	HF-R Adapter	No
ILPA Adapter	No	:IP Adapter	No	PSA Adapter	No

34-Button Aspire PoE IPhone - BK — P/N 0890073B

This 34-button IP keyset is a 802.3af PoE-compliant phone which provides a network connector, allowing it to be used with the VoIP feature.

The 34-Button Aspire IPhone has a 3-line, 24-character display with four interactive soft keys for intuitive feature access. In addition, it has 24 user-programmable function keys (with Dual LEDs) for one-button access to co-workers, features and outside lines. The telephone also provides 10 user-programmable One-Touch (Personal Speed Dial) keys and 15 additional fixed feature keys.

The 34-Button Aspire IPhone has a built-in speakerphone and can accept optional IP adapters (PSA, ADA2). You can also assign 110-Button DSS Consoles to these phones, but they must be on site (not networked). It provides Handsfree Answerback, Intercom voice-announcements. In addition, the telephone provides a built-in wall-mount bracket, as well as adjustable legs which allow each phone to be angled at a height which best suits the user.

✍ **At a glance**

34-Button Display Telephone - Part Number 0890073B

Function Keys:	✔	Accepts 110-Button DSS:	✔	Accepts 24-Button DLS:	No
Handsfree (Speakerphone):	✔	Dual LEDs:	✔	ADA Adapter:	No
ADA2 Adapter:	✔	APA Adapter:	No	APR Adapter:	No
CTA Adapter:	No	CTU Adapter:	No	HF-R Adapter	No
ILPA Adapter	✔	:IP Adapter	No	PSA Adapter	✔

H.323 IP Phone — P/N 780005

This UIP300 H.323 IP phone is a business IP phone in an enterprise LAN environment and will be connected to IP PBX systems via an RJ45 network cable.

Standard Telephone Features:

- Alphanumeric LCD display with 2 lines of 24 characters
- 10 LED indicators (Line 1 / Line 2[1] / Status / Mute / Speaker (Headset) / 5 Function keys)
- 12 Key Dial Pad
- 20 Specific Keys (5 Function keys [voice mail or personal speed dial], Menu, Select, Cancel/Del, Transfer, Mute, Redial, Hold, Conference, Speaker, Line 1 and 2 keys, Volume Up and Down keys, Menu Up and Down keys)
- Local Date and Time
- Call Duration Display
- Volume Control for Speaker, Handset, Headset, and Ringer
- Phone Book, Speed Dial, Dial from Call Logs (30 Outgoing Calls, 30 Incoming Calls and 15 Missed Calls)
- Redial, Hold[2], Mute
- Call Waiting, Call Forward, Call Transfer, 3-Way Conference, Do Not Disturb (DND)
- Display Caller ID (Name & Number)
- On-hook Dialing, Handsfree Talking (Full Duplex)
- DTMF Generation
- 8 Ringer Tones

This phone does not support the following:

- The phones cannot send digits after a call has been placed and before it is answered. This means that features which use single digit service codes, such as Voice Over and Barge-In, are not available with this type of phone.
- These phones do not provide P-codes, and therefore, cannot be used with the inDepth application.
- The Message Waiting/Voice Mail LED will not flash when there are new messages.

VoIP Specific Features:

- H.323 v1, 2 Standard Compliant
- Gatekeeper Routed and Direct Routed Call Models
- Voice Codec: G.711 (64kbit/s, u-Law and A-law), G723.1, G729AB
- E.164 Dialing
- Acoustic Echo Cancellation (G.167)
- Rapid Configuration with DHCP or Statically Configured IP Address
- Voice Activity Detection (VAD)
- QoS (IEEE 802.1 p/q Based and DiffServ)
- Jitter Compensation
- 10/100 Base-T Ethernet Interface

[1] Only one phone number will be assigned to this IP phone. Line 2 is not available for a gateway system.

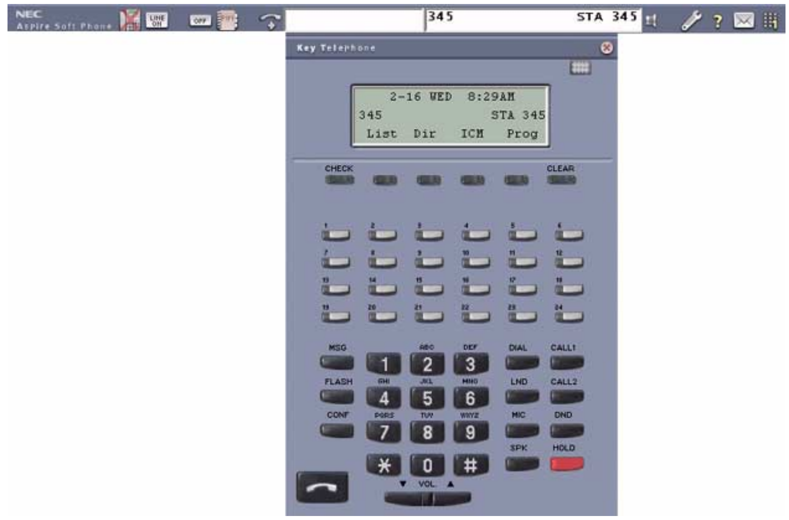
[2] Hold, Transfer, Call Forward and Conference will not be available in the IP address call mode but in the phone number dial mode only.

✍ **At a glance**

H.323 Telephone - Part Number 780005

Function Keys:(not Aspire keys)	✔	Accepts 110-Button DSS:	**No**	Accepts 24-Button DLS:	**No**
Handsfree (Speakerphone):	✔	Dual LEDs:	**No**	ADA Adapter:	**No**
ADA2 Adapter:	**No**	APA Adapter:	**No**	APR Adapter:	**No**
CTA Adapter:	**No**	CTU Adapter:	**No**	HF-R Adapter	**No**
ILPA Adapter	**No**	:IP Adapter	**No**	PSA Adapter	**No**

Aspire Soft Phone — P/N 0893641



Aspire Soft Phone is a business phone which works on a personal computer. It enables various telephony functions using an IP network connecting with the Aspire S/ Aspire system

This allows you to capitalize on the advantages of a converged voice and data network whether you're in the office or on the road. The Aspire Soft Phone application combines traditional business communication needs with the data applications you require.

The Aspire Soft Phone delivers high quality voice via a USB-connected handset or headset or via a PC sound card, microphone and speakers. The Aspire Soft Phone functions include, not only making and receiving calls, but also placing calls on hold, intercom calls, conferencing, etc. The application can display a layout of an Aspire keyset to allow for ease of operation. Using the cursor, simply point and click to operate the Aspire Soft Phone as you would an Aspire keyset.

With Soft Phone software 1.0.3.0 or higher and Aspire system software 4.93 or higher, a USB camera can be connected to the Aspire Soft Phone PCs, allowing a user to make a phone call to another user registered on the same system and provide a video transmission for the call (as long as both users have a USB camera connected and the Aspire Soft Phone application installed with a serial key providing the video function).

In-Line Power Adapter (ILPA-R) — P/N 780122

The In-Line Power Adapter (ILPA-R), which is IEEE 802.3af compliant, detects power from a PoE-compatible ethernet switch and passes it to the IP terminal. The ILPA does the negotiation and detection with the switch and then relays the power to the IP terminal device. This provides an additional way to power the NEC IP terminals (Aspire IPhone or Aspire Keyset with IP Adapter). With this adapter, the IP terminals on the Aspire can be powered using:

- Local power connecting the IP terminal to a local AC wall outlet using the AC-2R Adapter (P/N 780135)
- NEC power supply PoE-managed switch (BlueFire 200/24) (in-line and spare pair detection)
- Aspire 8SHUBU PCB (P/N 0891021) (spare pair detection)
- Cisco Data Switch - CDP supported (in-line and spare pair detection)
- In-Line Power Adapter

Keep the following in mind when installing an ILPA:

- Only IP telephones supported by center feed can be used.
- This adapter can not be used with the H.323 telephones.
- When center feed is used, first unplug the adapter from the ethernet switch before changing the SW1 setting on the back of the adapter.
- Please note that the ILPA-R adapter is intended for use with the Aspire IPhones (P/N 0890065) and IP Adapters (P/N 0890060). Installing any other device into the telephone port of the ILPA-R may result in damage to the device.
- When powering an IP phone using an ILPA-R adapter, the phone should not get connected to a port on the 8SHUBU PCB.

2

AC Adapter (AC-2R) — P/N 780135

The AC Adapter is required for the IP adapter or Aspire IPhone if external power is needed. Also required for APR, CTU, and Speakerphone optional keyset adapters.

Aspire (M/L and XL) Common Equipment

Aspire 8 Slot KSU w/o PS — P/N 0890000

The KSU is the system's control center. It houses the Power Supply, has nine PCB slots and provides for connection to trunks and extensions. The first slot in the KSU is dedicated to the NTCPU. The next slot is a universal slot which should be reserved for a Digital Station Card. The remaining seven slots are also universal. They can be used for any combination of Common, Trunk or Station PCBs. The KSU can be floor, wall or rack mounted.

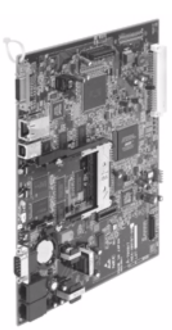
- You should plug a Digital Station Card into the first universal slot.

Aspire XL Power Supply Cabinet w/o PS — P/N 0890068

The Aspire XL power supply cabinet accommodates up to two AC/DC power supplies. This cabinet is to be installed as the bottom cabinet of a 3-cabinet system (the top two cabinets containing the NTCPU, DC/DC Converters, and PCBs).

Note: The Aspire XL system can support mixed hardware configurations. With an Aspire XL AC/DC power supply cabinet and a DC/DC Converter in one cabinet, the second cabinet can contain up to two Aspire M/L power supplies (P/N 0891000).

Central Processing Unit (NTCPU) PCB — P/Ns 0891002 & 0891038



The NTCPU controls all the functions and operations of the Aspire system using the system software loaded into the NTCPU memory. One 32-bit NTCPU PCB must be installed in the CPU slot in the Main Cabinet. There are two versions of NTCPUs. The first version, P/N 0891002, is a 64-port basic NTCPU. The second version, P/N 0891038, is a feature-enhanced, 256 extension port NTCPU.

To upgrade from the basic 64-port NTCPU, a Feature Upgrade chip (P/N 0891039) is available. The NTCPU provides a connector (CN14) for the upgrade PAL EPROM chip. Make sure when installing this upgrade chip on the NTCPU that you wear a grounded wrist strap. Using software 4.0E and higher, the Feature Upgrade PAL chip (P/N 0891039), supports 128 ports for trunks, extensions, and voice mail (internal and external). With prior software, only 64 ports are available.

The 64-port basic CPU (P/N 0891002), with the basic factory-installed PAL chip, provides:

- 64 ports maximum for trunks, extensions, voice mail (internal or external), Aspire Wireless 2.4 GHz and IP Phones

  With software 4.93+, if all 64 ports are required and IntraMail is not being used, set Program 47-01-17 to "0" to make these ports available.
- 256 virtual extensions
- Supports the 4VOIPU PCB and 4VOIPDB
- Supports TAPI 1.x
- VRS (***Requires DSPDB Daughter Board and software 2.00+*** - prior to this software version, the VRS is not supported by the basic factory-installed PAL chip)
- T1 Trunks (***Requires software 4.0E+*** - prior to this software version, this was not supported)
- PRI Trunks (Requires ***software 4.0E+*** - prior to this software version, this was not supported)
- DSPDB Daughter Board (providing 32 channels for the DTMF Receiver, Call Progress Tone Detection and Caller ID Receivers) (***Requires software 4.0E+*** - prior to this software version, the DSPDB did not support the additional channels)
- Supports the 32ESIU PCB
- inDepth without ACD (***Requires software 2.65+***)

2

NOT SUPPORTED by the 64-port basic CPU, with the basic factory-installed PAL chip:

- Expansion Cabinet
- Third-Party CTI/TAPI 2
- 16VOIPU PCB and 16VOIPDB
- ACD
- BRI S-Bus/T-Bus
- E&M Trunks
- Networking

The 64-port basic CPU (P/N 0891002), with the Feature Upgrade PAL chip, provides:

- 128 ports maximum for trunks, extensions, voice mail (internal or external), Aspire Wireless 2.4 GHz and IP Phones

  (***Requires software 4.0E+*** - prior to this software version, only 64 ports are supported)
- 256 virtual extensions
- Supports the 4VOIPU PCB and 4VOIPDB
- Supports TAPI 1.x
- VRS (Requires DSPDB Daughter Board)

The 64-port basic CPU, with the Feature Upgrade PAL chip, supports:

- Expansion Cabinet
- DSPDB Daughter Board (providing 32 channels for the DTMF Receiver, Call Progress Tone Detection and Caller ID Receivers)
- Third-Party CTI/TAPI 2
- 16VOIPU PCB and 16VOIPDB
- T1 Trunks
- ACD/inDepth
- PRI Trunks
- BRI S-Bus/T-Bus
- E&M Trunks
- Networking
- 32ESIU PCB

The enhanced CPU (P/N 0891038) provides:

- 200 trunk ports maximum
- 512 extension ports maximum
  - Aspire M/L: 256 analog/digital extensions,
  - Aspire XL: 384 analog/digital extensions
  - Aspire M/L or XL: NEC Wireless 2.4 GHz and IP Phones can have a maximum of 512 ports, however, more than 256 (Aspire M/L) or 128 (Aspire XL) Wireless and IP phones reduces the number of available extension ports.
- 256 virtual extensions
- Supports the 4VOIPU PCB and 4VOIPDB
- Supports TAPI 1.x
- VRS (Requires DSPDB Daughter Board)

The enhanced CPU supports:

- Expansion Cabinet
- DSPDB Daughter Board (providing 32 channels for the DTMF Receiver, Call Progress Tone Detection and Caller ID Receivers)
- Third-Party CTI/TAPI 2
- 16VOIPU PCB and 16VOIPDB
- T1 Trunks
- ACD/inDepth
- PRI Trunks
- BRI S-Bus/T-Bus
- E&M Trunks
- Networking
- 32ESIU PCB

Each version of the NTCPU provides the following:

- Five diagnostic LEDs which indicate the status of various system functions
  During normal operation, the "RUN" LED will be flashing and the remaining LEDs will be off.
- 1019x1019 Time Division Multiplex Switch (TDM Switch)
- Digital Phase Locked Loop (DPLL)
- Tone Generator
- DTMF Tone Sender
- 32 Tone Resources (for DTMF Receiver, Caller ID Receiver, and Call Progress Tone Detection)
- System Tone Sender
- MFC Tone Sender
- MF Signal Sender (Sends caller information to CO for E911)
- Call Progress Tone Detector
- C-Channel Control
- Conference; 64 Channels
- Caller ID Receiver; 32 Channels
- Caller ID Sender; 4 or 10 Channels for Analog Stations
  This can be expanded up to 20 by disabling 32 channels of the Conference circuits and disabling the MFC Tone Sender.
- A reset switch (RES) which can be used to reset the system
- A load switch (LOAD) which is used for initial system startup or when upgrading system software
- One Serial Port
- One USB Port (requires USB driver - download from NEC web site)
- One Ethernet Port (10 Base-T/100 Base-TX)
- One PCMCIA Slot
- One EXIFU Interface Connector
- Two Audio Input Terminals

- One Audio Output Terminal
- One Night Mode Terminal for External Switch
- One Music On Hold External Source
- HDLC Packet Processing
- Real Time Clock (tolerance 30 seconds/month)
- Internal MOH Generation
- One Connector for DSPDBU Daughter Board
- One Connector for PAL EPROM
- One lithium battery (Sony CR2032 or equivalent) which provides battery back-up of system data and RAM memory for approximately 30 months

! IMPORTANT!
After removing a previously installed NTCPU, handle the PCB, carefully, from the edges. If certain solder points/resistors are touched on the back of the PCB, some RAM/temporary memory may be lost (ex: time, date, user-defined settings, etc.)

2

Aspire M/L Power Supply — P/N 0891000

The Power Supply provides the DC voltage for the Cabinet PCBs and all telephones connected to the Cabinet Station PCBs.

Each Aspire M/L system cabinet must have at least one power supply installed. In order to determine if a second power supply is required, refer to the load factor charts located in Section 2 of the Aspire Hardware Manual (P/N 0893100).

Note: One power supply can provide power to 64 analog or digital telephones. If more than 64 telephones are connected to a cabinet, a second power supply must be used.

Aspire XL Power Supply Set — P/N 0890069

This set contains a power supply and DC/DC Converter required to convert an Aspire M/L system to an Aspire XL system, with expanded physical port capacity.

This kit includes the following:

- **AC/DC Power Supply (IP1WW-PSADU-A1) for the Aspire XL Power Supply Cabinet - P/N 0892011**

  The Power Supply provides the DC voltage to the DC/DC Converter which, in turn, powers the PCB cabinet and all telephones connected to the station PCBs.

  Up to two AC/DC Power Supplies maximum can be installed per Power Supply Cabinet. When two AC/DC Power Supplies are installed, two separate AC cords will each require an AC outlet connection from the bottom Power Supply Cabinet. Use the power cords included with the Aspire system cabinet(s) for connecting the power supplies to the AC outlet. The Aspire system cabinet is no longer directly connected to an AC outlet with the AC/DC Power Supplies installed.

Note: A multi-cabinet system can use different power supplies (0891000 and 0892011), however, if a cabinet has more than 128 -48V ports, the AC/DC power supply and DC/DC Converter is required.

If both AC/DC power supplies and DC/DC Converters are installed, then a maximum of 384 ports (greater than 128 ports in each cabinet) are supported by -48V output.

- **DC/DC Converter (IP1WW-PSDDU-A1) for the Aspire XL System Cabinet - P/N 0892012**

  A maximum of two DC/DC Converters are installed per system - one in each system cabinet.

  A DC/DC Converter replaces the two power supplies (P/N 0891000) used in the system cabinet of the Aspire M/L system.

  If an Aspire M/L system is upgraded to an Aspire XL, the existing power supplies will no longer be required as they are replaced by the DC/DC Converter.

  Note: A multi-cabinet system can use different power supplies (0891000 and 0892011), however, if a cabinet has more than 128 -48V ports, the AC/DC power supply and DC/DC Converter is required.

  If both AC/DC power supplies and DC/DC Converters are installed, then a maximum of 384 ports (greater than 128 ports in each cabinet) are supported by -48V output.

- **Cable for Power (connects the AC/DC Unit and DC/DC Unit) - P/N 0892013**

  This cable allows for a power connection from one AC/DC Power Supply in the bottom cabinet to one DC/DC Converter in the system cabinet.

  Note: The power cord for the system cabinet (located on the back of the cabinet) is no longer required with this cable connection.

- **Cable for Signal (connects the AC/DC Unit and DC/DC Unit) - P/N 0892010**

  This cable allows for a signal connection from one AC/DC Power Supply in the bottom cabinet to one DC/DC Converter in the system cabinet.

KSU Expansion Set — P/N 0891001

This set is required to connect a second cabinet, allowing expansion from an 8 slot to a 16 slot system. It consists of an EXIFU PCB, EXIFU Cable and Joint Plate.

19” Rack Mount Bracket — P/N 0891300

The 19” Rack Mount Bracket is required to install the KSU onto a 19” rack mounting system. If rack-mounting all three cabinets of an Aspire XL system, the power supply cabinet (P/N 0890068) must be mounted on a separate bracket due to the bracket weight requirements (150 lbs maximum).

Feature Upgrade for 0891002 — P/N 0891039

The Feature Upgrade for the 0891002 is added to the 64 Port Basic CPU (0891002). It provides such features as 16 channel media gateway cards, ACD, T1/PRI, E&M, etc. Refer to the NTCPU info on **Central Processing Unit (NTCPU) PCB — P/Ns 0891002 & 0891038** (page 32) for more.

With software 4.0E or higher, the Feature Upgrade PAL chip can support 128 ports for trunks, extensions, and voice mail (internal and external). With prior software, only 64 ports are available.

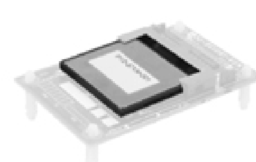
DSP Resource Daughter Board (DSPDB) — P/N 0891003

The DSPDB provides additional DSP resources as well as the option for the VRS (Voice Response System) or Voice Mail features. This daughter board is mounted on the NTCPU and provides:

- 32 Tone Resources (for DTMF Receiver, Caller ID Receiver, and Call Progress Tone Detection)
- 16 VRS Circuits with a VRS Flash Card Installed (replays up to 16 circuits simultaneously, recording: up to 8 circuits simultaneously)
- Compact Flash Slot for VRS Features

The receiver circuits are used for DTMF receivers, call progress tone detection, and Caller ID receivers.

VRS Compact Flash (DSPDB Compact Flash Card) (DSPDB) — P/N 0891040



The VRS Flash Card adds Voice Announce/Automated Attendant features and is installed on the DSP Resource daughter board.

Aspire Trunk PCBs

4 CO Loop Start Trunk Card (4COIU-LS1) — P/N 0891005

There are two different types of the COIU PCB. One providing ground start trunks - the other is for loop start trunks only.

The 4COIU-LS1 PCB provides:

- 4 analog loop start line/trunk circuits - no ground start is provided
- 4 trunk status LEDs
- 4 Caller ID Circuits
- 2 Power Failure Transfer Circuits
- 1 PCB status LED
- 1 run/block switch

The CN3 connector provides connection to 4 analog trunk ports, ***which are polarity sensitive (tip to tip, ring to ring)***. The power failure circuits, however, are not polarity sensitive. A maximum of 15 4COIU-LS1 PCBs per system is allowed, each PCB consuming 4 ports.

8 CO Loop Start Trunk Card (8COIU-LS1) — P/N 0891004

The 8COIU-LSI provides the same features as the 4COIU-LS1 except for the following changes:

The 8COIU-LS1 PCB provides:

- 8 analog loop start line/trunk circuits - no ground start is provided
- 8 trunk status LEDs
- 8 Caller ID Circuits
- 2 Power Failure Transfer Circuits
- 1 PCB status LED
- 1 run/block switch

The CN3 and CN5 connectors each provide connection to 4 analog trunk ports, ***which are polarity sensitive (tip to tip, ring to ring)***. The power failure circuits, however, are not polarity sensitive. A maximum of 15 8COIU-LS1 PCBs per system is allowed, each PCB consuming 8 ports.

4 CO Loop Start/Ground Start Trunk Card (4COIU-LG1) — P/N 0891029

There are two different types of the COIU PCB. One providing ground start trunks - the other is for loop start trunks only.

The 4COIU-LG1 PCB provides:
- 4 analog ground start / loop start trunk circuits
- 4 trunk status LEDs
- 4 Caller ID Circuits
- 2 Power Failure Transfer Circuits
- 1 PCB status LED
- 1 run/block switch

The CN3 connector provides connection to 4 analog trunk ports, ***which are polarity sensitive (tip to tip, ring to ring)***. The power failure circuits, however, are not polarity sensitive. A maximum of 15 4COIU-LG1 PCBs per system is allowed, each PCB consuming 4 ports.

! Important!
- When using the COIU-LG1 PCB for ground start trunks, the PBX ground ***must*** be connected or the trunks will not function correctly.
- When connecting the RJ61 cables to the COIU PCB, note the position of the Power Failure connector. Do not confuse this connector as the trunk connector.

8 CO Loop Start/Ground Start Trunk Card (8COIU-LG1) — P/N 0891028

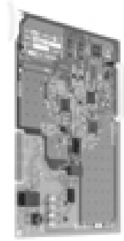
The 8COIU-LG1 provides the same features as the 4COIU-LG1 except for the following changes:

The 8COIU-LG1 PCB provides:
- 8 analog ground start / loop start trunk circuits
- 8 trunk status LEDs
- 8 Caller ID Circuits
- 2 Power Failure Transfer Circuits
- 1 PCB status LED
- 1 run/block switch

The CN3 and CN5 connectors each provide connection to 4 analog trunk ports, ***which are polarity sensitive (tip to tip, ring to ring)***. The power failure circuits, however, are not polarity sensitive. A maximum of 15 8COIU-LG1 PCBs per system is allowed, each PCB consuming 8 ports.

! Important!
- When using the COIU-LG1 PCB for ground start trunks, the PBX ground ***must*** be connected or the trunks will not function correctly.
- When connecting the RJ61 cables to the COIU PCB, note the position of the Power Failure connector. Do not confuse this connector as the trunk connector.

T1/PRI Interface Card (1PRIU) — P/N 0891009



For T1 and ISDN Primary Rate Interface (PRI) applications, install a T1/PRI Interface PCB. This PCB has a single 24-channel 64Kb/s digital signal circuit which can be configured for either T1 trunks or PRI. Each PCB connects to the network via an NTI Network Termination.

If set for T1, the T1/PRI PCB gives the system 24 trunks in a single universal slot. These trunks can be one of the following:

- Loop Start
- Ground Start
- DID
- E&M Trunks
- ANI/DNIS E&M Trunks

T1 gives the system the advantages of advanced digital trunking as well as conserving universal slots. For example, a system with 12 loop start trunks, two tie lines and six DID trunks would use up five universal slots. With T1 all these trunks would be available in a single universal slot, freeing up four additional universal slots for other uses.

If set for PRI, each T1/PRI PCB provides 24 PRI (23 B& 1 D) channels running at 1.544Mbps with 64Kb/s clear channel. The PCB supports the following PRI services:

- Basic PRI Call Control (BCC)
- Display of incoming caller's name and number (when allowed by the telco)
- Speech and 3.1 KHz audio

When installed, the T1/PRI Interface PCB uses the first available block of 24 consecutive trunks. For example, if you have an COIU PCB installed for trunks 1-8, the T1/PRI Interface PCB will automatically use trunks 9-32. If you have COIU PCBs installed for trunks 1-8 and 17-24, the T1/PRI PCB will use trunks 25-48. The T1/PRI Interface PCB cannot use trunks 9-16 (even if available) since they are not part of a consecutive block of 24 trunks.

The T1/PRI PCB requires one universal slot and provides a Block switch to busy out the PCB. When used for T-Bus, up to 8 PCBs can be installed in the system. When used as S-Bus, up to 10 PCBs can be installed.

4 DID/OPX (4DIOPU) Card — P/N 0891013

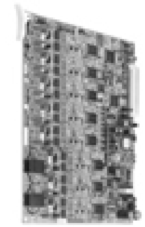
The 4DIOPU PCB supports the analog DID and single line telephone interface functions (such as Off-Premise Extension). The function type is assigned in programming for each port. The circuit types, however, should be grouped together. For example, with 3 DID circuits and 1 OPX circuit, they should be grouped as DID, DID, DID and OPX and not DID, DID, OPX and DID.

The DIOPU PCB provides:

- 4 DID trunk circuits
- 4 DID trunk status LEDs
- 1 PCB status LED
- 1 run/block switch

The CN3 connector provides connection to 4 analog DID trunk ports, ***which are polarity sensitive (tip to tip, ring to ring)***. The OPX circuits, however, are not polarity sensitive. The DIOPU requires one universal slot, with 15 maximum PCBs per system. When used only as DID, the PCB consumes 4 trunk ports. If OPX trunks are defined in Program 10-03-01 for the PCB, then 4 station ports are assigned as well.

8 DID/OPX (8DIOPU) Card — P/N 0891012

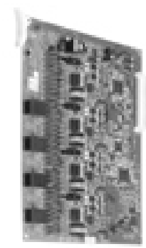


The 8DIOPU PCB supports the analog DID and single line telephone interface functions (such as Off-Premise Extension). The function type is assigned in programming for each port. The circuit types, however, should be grouped together. For example, with 3 DID circuits and 1 OPX circuit, they should be grouped as DID, DID, DID and OPX and not DID, DID, OPX and DID.
The DIOPU PCB provides:

- 8 DID trunk circuits
- 8 DID trunk status LEDs
- 1 PCB status LED
- 1 run/block switch

The CN3 and CN5 connectors each provide connection to 4 analog DID trunk ports, ***which are polarity sensitive (tip to tip, ring to ring)***. The OPX circuits, however, are not polarity sensitive. The DIOPU requires one universal slot, with 15 maximum PCBs per system. When used only as DID, the PCB consumes 8 trunk ports. If OPX trunks are defined in Program 10-03-01 for the PCB, then 8 station ports are assigned as well.

4 E&M Tie Line Card (4TLIU) — P/N 0891011



The 4TLIU Tie Line PCB is an out band dial type analog tie line interface PCB. The PCB supports system connections to either 2-wire (four lead, tip/ring) or 4-wire (eight lead, tip/ring/tip 1/ring 1) E&M signalling tie lines (determined in Program 10-03). Using switches on the PCB, each circuit type can be set as Type I, II, III, IV, or V. Each PCB provides:

- 4 4-circuit tie line interfaces
- 4 tie line status LEDs
- 1 PCB status LED
- 1 run/block switch
- 2 straps and 1 switch per circuit to determine the circuit type

A maximum of 15 PCBs per system are allowed, providing 60 tie line trunks and it can be plugged into any universal slot. Each PCB consumes 4 ports.

2 BRI Card (2BRIU) — P/N 0891006

The 2BRIU provides:

- 2 2-Channel Circuits (2B + D) configured as T-Bus or S-Bus
- 64 Kb/s Clear B-Channel and 16 Kb/s D-Channel
- 2 trunk/extension status LEDs
- 1 PCB status LED
- 1 run/block switch

The BRI Interface PCB uses a single universal slot. A maximum of 15 2BRIU PCBs can be installed. When used only as T-Bus, the PCB consumes 4 trunk ports. If S-Bus is defined in Program 10-03-01 for the PCB, then 4 station ports are assigned as well. Each PCB connects to the network via an NTI Network Termination. With the maximum number of PCBs installed, the following can be provided:

- The 2BRI provides 30 BRI circuits and 60 BRI channels.

The trunk circuit can be connected to either an ISDN trunk or ISDN telephone set, depending on the SW102 through SW202 switch settings. When used for S-Bus, a maximum of 8 ISDN terminals can be connected to each circuit.

Two ISDN telephone circuits (1-2) are supplied with DC power from the Aspire system.

4 BRI Card (4BRIU) — P/N 0891007

The 4BRIU PCB provides:

- 4 2-Channel Circuits (2B + D) configured as T-Bus or S-Bus
- 64 Kb/s Clear B-Channel and 16 Kb/s D-Channel
- 4 trunk/extension status LEDs
- 1 PCB status LED
- 1 run/block switch

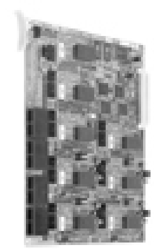
The BRI Interface PCB uses a single universal slot. A maximum of 15 4BRIU PCBs can be installed. When used only as T-Bus, the PCB consumes 8 trunk ports. If S-Bus is defined in Program 10-03-01 for the PCB, then 16 station ports are assigned as well. Each PCB connects to the network via an NTI Network Termination. With the maximum number of PCBs installed, the following can be provided:

- The 4BRI provides 60 BRI circuits and 120 BRI channels.

The trunk circuit can be connected to either an ISDN trunk or ISDN telephone set, depending on the SW102 through SW402 switch settings. When used for S-Bus, a maximum of 8 ISDN terminals can be connected to each circuit.

Four ISDN telephone circuits (1-4) are supplied with DC power from the Aspire system.

8 BRI Card (8BRIU) — P/N 0891008



The 8BRIU PCB provides:

- 8 2-Channel Circuits (2B + D) configured as T-Bus or S-Bus
- 64 Kb/s Clear B-Channel and 16 Kb/s D-Channel
- 8 trunk/extension status LEDs
- 1 PCB status LED
- 1 run/block switch

The BRI Interface PCB uses a single universal slot. With the 8BRIU PCBs, a maximum of 12 PCBs can be installed when used for T-Bus connections - 15 PCBs can be installed when used for S-Bus connections.When used only as T-Bus, the PCB consumes 16 trunk ports. If S-Bus is defined in Program 10-03-01 for the PCB, then 32 station ports are assigned as well. Each PCB connects to the network via an NTI Network Termination. With the maximum number of PCBs installed, the following can be provided:

- The 8BRI, when used as T-Bus, provides 96 BRI circuits and 192 BRI channels. When used as S-Bus, 120 BRI circuits and 240 S-Bus station ports are provided.

The trunk circuit can be connected to either an ISDN trunk or ISDN telephone set, depending on the SW102 through SW802 switch settings. When used for S-Bus, a maximum of 8 ISDN terminals can be connected to each circuit.

The first 4 ISDN telephone circuits (1-4) are supplied with DC power from the Aspire system. If the last four circuits (5-8) are to be used for S-Bus, they must use ISDN telephone sets which provide their own local power supply as the system does not provide DC power to these circuits.

Aspire Station PCBs

32 Aspire Digital Card (32ESIU) — P/N 0891058

The 32ESIU PCB provides:
- 32 digital extension circuits (used for digital telephones, DSS consoles, 1SLTAD adapters, 2PGDAD adapters)
- 8 extension status LEDs (each LED indicates status for 4 extensions - BL1 used for ports 1-4, BL2 for ports 5-8, etc.).
- 1 PCB status LED
- 1 run/block switch
- B2 channel on the first 4 ports on each PCB (the APR(B2 mode) adapter or PGDAD module must be installed on one of these first 4 ports)

The 32ESIU PCB has four connectors (CN2, CN3, CN4, CN5). Each connector provides two modular jacks which are each used to connect up to four Aspire digital extensions. The PCB consumes 32 ports, ranging from ports 001-384.

In order to program the system, an 8ESIU, 16ESIU or 32ESIU PCB should be installed in slot 1. However, system programming can be done using the PCPro or WebPro applications or through a VoIP telephone. The 32ESIU requires one universal slot, with a maximum of 8 per Aspire M/L system (4 max. per cabinet) or 12 PCB's per Aspire XL system (8 max. per cabinet). When more than 4 ESIU PCBs (128 ports) are installed in a cabinet, a Power Supply set is required (P/N 0890069).

16 Aspire Digital Card (16ESIU) — P/N 0891014

The 16ESIU PCB provides:
- 16 digital extension circuits (used for digital telephones, DSS consoles, 1SLTAD adapters, 2PGDAD adapters)
- 4 extension status LEDs (each LED indicates status for 4 extensions - BL1 used for ports 1-4, BL2 for ports 5-8, BL3 for ports 9-12, and BL4 for ports 13-16).
- 1 PCB status LED
- 1 run/block switch

The CN102, CN103, CN202, and CN203 connectors each provide connection to 4 digital station ports. The PCB consumes 16 ports, ranging from ports 001-256.

In order to program the system, an 8ESIU or 16ESIU PCB should be installed in slot 1. However, system programming can be done using the PCPro or WebPro applications or through a VoIP telephone. The ESIU requires one universal slot, with a maximum of 16 PCB's per system.

8 Aspire Digital Card (8ESIU) — P/N 0891015

The 8ESIU PCB provides:
- 8 digital extension circuits (used for digital telephones, DSS consoles, 1SLTAD adapters, 2PGDAD adapters)
- 2 extension status LEDs (each LED indicates status for 4 extensions - BL1 used for ports 1-4, BL2 for ports 5-8)
- 1 PCB status LED
- 1 run/block switch

The CN102 and CN103 connectors each provide connection to 4 digital station ports. The PCB consumes 8 ports, ranging from ports 001-256.

In order to program the system, an 8ESIU or 16ESIU PCB should be installed in slot 1. However, system programming can be done using the PCPro or WebPro applications or through a VoIP telephone. The ESIU requires one universal slot, with a maximum of 16 PCB's per system.

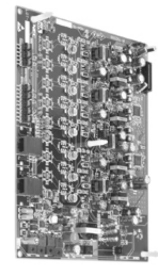
8 Analog Station Card (8SLIU) — P/N 0891017

The 8SLIU PCB provides:
- 8 analog extension ports (used for on-premise analog telephones, fax machines, and analog modems)
- 8 extension status LEDs
- 1 PCB status LED
- 1 run/block switch
- Connector for 8SLIDB Daughter Board
- Ring Generator
- 8 SW1 switches which provide constant current type battery feeding (set to either 20mA [default] or 35mA)
- Message Wait Lamping Ability

Note: When connecting a fax machine or analog modem, make sure to set Program 15-03-03 to '1' (special terminal) to avoid communication problems.

The CN3 and CN5 connectors each provide connection to 4 analog station ports and are not polarity sensitive. The 8SLIU is installed in a universal slot, with 15 maximum per system. The PCB consumes 8 ports, ranging from ports 001-256.

8 Analog Station Expansion Daughter Board (8SLIDB) — P/N 0891018



The 8SLIDB daughter board provides:

- 8 analog extension ports (used for on-premise analog telephones, fax machines, and analog modems)
- 8 SW1 switches which provide constant current type battery feeding (set to either 20mA [default] or 35mA)
- Connector for 8SLIU PCB
- Ring Generator
- Message Wait Lamping Ability

Note: When connecting a fax machine or analog modem, make sure to set Program 15-03-03 to '1' (special terminal) to avoid communication problems.

The CN3 and CN5 connectors each provide connection to 4 analog station ports and are not polarity sensitive. The 8SLIDB is installed on the 8SLIU PCB, with 15 maximum per system. The daughter board consumes 8 ports, ranging from ports 001-256.

2

4 Aspire Wireless (DECT) Station PCB (4DSIU) — P/N 0891090

The 4DSIU PCB provides the ability to use Aspire Wireless (DECT) phones with the Aspire system. Each 4DSIU PCB allows up to 4 base stations to be connected. Each RFP connector on the PCB provides connection to 4 Base Stations, using 4 station ports (ranging from 002-512 if manually programmed or 257-512 if automatically selected). As the Base Stations are powered by the DSIU PCB, whenever a DSIU PCB is installed in an Aspire cabinet, two power supplies must be installed. Only one DSIU PCB can be installed in a system with a maximum of 120 handsets connected. This PCB does not consume any ports - the ports are assigned as the telephones are registered to the system.

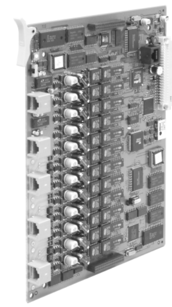
The maximum number of B-channels supported is 32, which means that the DSIU PCB (any version) can support a maximum of 32 conversations at one time.

8 Aspire Wireless (DECT) Station PCB (8DSIU) — P/N 0891091

The 8DSIU PCB provides the ability to use Aspire Wireless (DECT) phones with the Aspire system. Each 8DSIU PCB allows up to 8 base stations to be connected. Each RFP connector on the PCB provides connection to 4 Base Stations, using 4 station ports (ranging from 002-512 if manually programmed or 257-512 if automatically selected). As the Base Stations are powered by the DSIU PCB, whenever a DSIU PCB is installed in an Aspire cabinet, two power supplies must be installed. Only one DSIU PCB can be installed in a system with a maximum of 120 handsets connected. This PCB does not consume any ports - the ports are assigned as the telephones are registered to the system.

The maximum number of B-channels supported is 32, which means that the DSIU PCB (any version) can support a maximum of 32 conversations at one time.

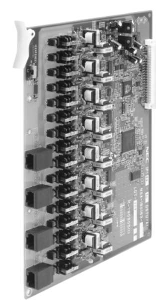
12 Aspire Wireless (DECT) Station PCB (12DSIU) — P/N 0891092



The 12DSIU PCB provides the ability to use Aspire Wireless (DECT) phones with the Aspire system. Each 12DSIU PCB allows up to 12 base stations to be connected. Each RFP connector on the PCB provides connection to 4 Base Stations, using 4 station ports (ranging from 002-512 if manually programmed or 257-512 if automatically selected). As the Base Stations are powered by the DSIU PCB, whenever a DSIU PCB is installed in an Aspire cabinet, two power supplies must be installed. Only one DSIU PCB can be installed in a system with a maximum of 120 handsets connected. This PCB does not consume any ports - the ports are assigned as the telephones are registered to the system.

The maximum number of B-channels supported is 32, which means that the DSIU PCB (any version) can support a maximum of 32 conversations at one time.

16 i-Series Phone PCB (16DSTU) — P/N 0891016



The i-Series Phone Card adds to the Aspire system the ability to connect up to 16 i-Series telephones with each port supporting 1 B-channel. The 16DSIU is installed in a universal slot, with 15 maximum per system. The PCB consumes 16 ports, ranging from ports 001-256. The following i-Series phones are compatible with the Aspire system:

Model 2 922xx/926xx Series Keysets
- 32-Button Display Phone, P/N 92293B / 92293W / 92673
- 32-Button Standard Phone, P/N 92290B / 92290W / 92670
- 16-Button Display Phone, P/N 92373C / 92373W / 92573 / 92563
- 16-Button Standard Phone, P/N 92370B / 92370W / 92570 / 92560

Model 3 i-Series Keysets
- 34-Button Display Phone, P/N 92783
- 28-Button Display Phone, P/N 92763
- 28-Button Standard Phone, P/N 92760
- 22-Button Display Phone, P/N 92753A / 92750A

In addition, the following hardware is also supported using DSTU ports:
- Remote Extender

The following i-Series hardware is **NOT** supported on the Aspire system:

- 2OPX
- 900/900i/910i Cordless Phone
- DCI-L
- DSLT
- VAU
- Data Module
- Speakerphone Module
- 3ACI
- DCI-A/B
- Digital VANGARD Voice Mail
- DSS Consoles
- Analog Module
- Off-Hook Voice Announce Module
- Super Display Phones

When using i-Series phones on the Aspire system, the following features are **NOT** supported:

- Directory Dial
- Soft Keys
- Changing Incoming CO and ICM Ring Tones (Program 11-11-20)
- Program 10-03 : PCB Setup (DSTU PCB has no programmable options)
- Program 90-07-01 : Extension Control
- Super Display Operation
- Headset Key
- Telephone System Programming (#*#*)
- Program 90-17-01 : Display Firmware Version

In addition to the above, the 92290x, 92670, 92370x, 92570, 92560, and 92760 phones do **NOT** support:

- Check Abandon Calls (CHECK + CALL2)
- Name Program (Service Code 800)
- Time and Date Display Modes (Program 20-02-07)
- Check Port/Name (CHECK + CALL1)
- Language Display (Program 15-02-01)

Aspire Optional Feature Equipment

Aspire Wireless (DECT)

Aspire Wireless (DECT) Base Station — P/N 780136

The Base Station, also known as the Radio Fixed Part (RFP), provides the link from the Aspire Wireless phone to the Aspire system. The Base Station is connected to the DSIU PCB using standard two wire (twisted pair) telephone cable, CAT 4 or CAT 5. No local power is required for the Base Station as it receives it power from the Aspire system. Each Base Station supports 4 simultaneous Aspire Wireless traffic channels at 32 Kbit/s.

Aspire Wireless (DECT) Repeater — P/N 780138

The Repeater, also known as the Wireless Radio Fixed Part (WRFP), provides extended coverage for low traffic areas not covered by a Base Station. Sufficient coverage for the main areas should be provided by the Base Stations. An external antenna can also be connected which extends the coverage area (maximum distance 3280’). Local power is required for the Repeater and it must be synchronized with a Base Station in the zone in which it will be providing coverage. Each Repeater can support 2 simultaneous Aspire Wireless traffic channels at 32 Kbit/s. As the Repeater is paired with the Base Station, these are not additional channels, but are available to handle calls from the Base Station as the user moves out the range from the Base Station to the Repeater’s area. The Repeater looks similar to the Base Station except that the modular jack on the back is used for connecting the local power source and there is a small plastic punch-out which provides the external antenna connection.

Aspire Wireless (DECT) External Antenna — P/N 780145

The external antenna is connected to the antenna connection on the back of the Repeater . When installing the External Antenna, direct it towards the Base Station to which the Repeater is paired. This external antenna can extend the range to a base station up to 3280 feet.

Aspire Wireless (DECT) Deployment Kit — P/N 780146

The Deployment Kit consists of two 2.4GHz Aspire Wireless handsets, a special base station, and connecting cables all packed into a handy hard-shell briefcase. Before installing the Aspire Wireless equipment, the deployment and diagnostics tool can be used to find the best place for the Base Stations by measuring the area of coverage of a Base Station within the building. Proper placement for the equipment is required for complete Aspire Wireless phone coverage.

In a building which has multiple telephone systems installed, the Deployment Tool can see all the systems to determine if there are too may systems running (too much radio activity in the air). It can be set to find all current stations running, but it is possible, by making a selection of the RFPI number, to search for a particular group of bases. If you wish to check the range of a particular base in a multi-base system, the Deployment Tool is able to lock onto the Base Station to avoid handover to any other units. You can also use the Deployment Tool to see channels with noise levels from -90 dBm to -60 dBm in jumps of 5 dBm.

Aspire Wireless (DECT) Repeater Programming Kit — P/N 780142

The Repeater is synchronized with the Base Station using the Repeater Programming Kit. This kit provides a cable with an RS232 connection which is used to connect the Repeater to a PC. A Windows application is then used to define the Base Station with which the Repeater should be synchronized.

Aspire Wireless (DECT) Service Tool Kit — P/N 780143

The Service Tool Kit contains a uniquely configured serial cable and a special charging unit. In addition to these items, the latest version of the Service Tool program is required. This can be downloaded from the Downloads page within the NEC Technical Support web site (http://ws.necii.com).

These items are used to set up special functions. When connected to a handset, the volume can be adjusted, the start-up text can be altered, new flash code and menus and be loaded, and the gain levels of the microphone and loudspeaker for the handset can be adjusted.

Conference Bridge

16 Port Conference Bridge - P/N 0891069

The 16CNFU PCB is a conference bridge system designed for the Aspire M/L/XL system. The PCB is installed in the telephone system and allows up to 16 parties to take part in a conference call. The participants dial a pre-assigned phone number at the determined time, optionally enter a password, and are prompted to speak their name which will be announced to the other conference participants. The 16CNFU PCB functionalities include:

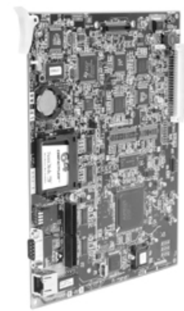
- Password protection is provided for each conference.
- Applicable voice messages and announcements (e.g., entry, password request, exit) are available.
- E-mail notification, when enabled, requires the organizer to enter the E-mail address of each participant to be sent notification of a pending conference. This option is selectable when setting up new conferences.

  The 16CNFU E-mail configuration supports SMTP mail server ONLY.
- Host Required, when enabled, requires the host/organizer to be logged into the conference before any other participant can enter. This option is selectable when setting up new conferences.
- Admission Control, when enabled, requires the organizer to dial a digit allowing each participant to enter the conference. This option is predefined access code.
- One customer greeting can be recorded for each 16CNFU PCB. Predefined password is necessary to record personal greetings.
- Password protection option for each conference.
- Remote conference programming with conference scheduler (via a Web User Interface).
- Programmable gain adjustments.
- Support for DTMF detection for manual setup options (Telephone User Interface).
- HTTP Interface for conference schedule management and conference PCB administration.

Note that when multiple 16CNFU PCBs are installed, the users in a conference call must all be on the same 16CNFU PCB. A call cannot span multiple PCBs.

Voice Mail

4 Port Aspire Mail (4FMS+) — P/N 0891052



Aspire Mail is a fully integrated, PCB-based “in-skin” Voice Mail with Automated Attendant. Its robust feature set rivals the capabilities of stand-alone products on a single, plug-in PCB. Using an on-board hard disk, Aspire Mail provides 4 voice mail ports, 200 mailboxes, and approximately 30 hours/7000 messages maximum of message storage.

Aspire Mail is programmable from a PC running a Windows™-based Admin program. The Admin PC connects locally to the Aspire Mail serial port or LAN connection. Remote programming and maintenance is available through either the LAN connection or the Aspire Mail built-in modem.

You can install 1 maximum Aspire Mail per system, with the PCB using 4 ports.

4 Port Aspire Mail DMS (4VMSU) — P/N 0891030

Aspire Mail DMS is a fully integrated, PCB-based “in-skin” Voice Mail with Automated Attendant. Its robust feature set rivals the capabilities of stand-alone products on a single, plug-in PCB. Using an on-board hard disk, Aspire Mail DMS provides 4 voice mail ports, 2000 mailboxes, approximately 1400 hours (min.)/14,000 messages (max.) of message storage, and 25 seats of Desktop messaging and Call Control (205 seats max).

Aspire Mail is programmable from a PC running a Windows™-based Admin program. The Admin PC connects locally to the Aspire Mail Plus LAN connection. Remote programming and maintenance is available through a LAN connection or the Aspire Mail built-in modem.

You can install 1 maximum Aspire Mail per system, with the PCB using 4 ports.

8 Port Aspire Mail DMS (8VMSU) — P/N 0891031

Aspire Mail DMS is a fully integrated, PCB-based “in-skin” Voice Mail with Automated Attendant. Its robust feature set rivals the capabilities of stand-alone products on a single, plug-in PCB. Using an on-board hard disk, Aspire Mail DMS provides 8 voice mail ports, 2000 mailboxes, approximately 1400 hours (min.)/14,000 messages (max.) of message storage, and 25 seats of Desktop messaging and Call Control (205 seats max).

Aspire Mail is programmable from a PC running a Windows™-based Admin program. The Admin PC connects locally to the Aspire Mail Plus LAN connection. Remote programming and maintenance is available through a LAN connection or the Aspire Mail built-in modem.

You can install 1 maximum Aspire Mail per system, with the PCB using 8 ports.

2-to-4 Port Aspire Mail Upgrade — P/N 0891044

The 2-to-4 Port Aspire Mail Upgrade (P/N 0891044) is a software upgrade that allows you to expand your Aspire Mail system (P/N 0891032) from 2 to 4 ports. This upgrade ***does not*** provide additional mailboxes or message storage.

4-to-8 Port Aspire Mail FMS+ Upgrade — P/N 0891066

The 4-to-8 Port Aspire Mail FMS+ Upgrade (P/N 0891066) is a software upgrade that allows you to expand your Aspire Mail FMS+ system (P/N 0891052) from 4 to 8 ports. This upgrade ***does not*** provide additional mailboxes or message storage.

4-to-8 Port Aspire Mail DMS Upgrade — P/N 0891010

The 4-to-8 Port Aspire Mail DMS Upgrade (P/N 0891010) is a software upgrade that allows you to expand your Aspire Mail DMS system (P/N 0891030) from 2 to 4 ports. This upgrade ***does not*** provide additional mailboxes or message storage.

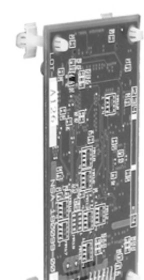
8 Port Aspire Mail DMS Expansion Daughter Board — P/N 0891036

The 8 Port Aspire Mail DMS Expansion Daughter Board (P/N 0891036) allows you to expand your Aspire Mail DMS system (P/N 0891030 or 0891031) by 8 ports. This upgrade ***does not*** provide additional mailboxes or message storage.

10 Seat Aspire Mail DMS Expansion License — P/N 0891020

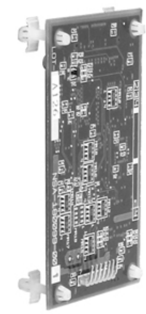
The 10 Seat Aspire Mail DMS Expansion License (P/N 0891020) allows you to add 10 seats of Desktop Messaging/Call Control to your Aspire Mail DMS system (P/N 0891030 or 0891031). A maximum of 205 seats are possible.

4 Port Aspire Mail Plus Expansion Daughter Board (4VMDB) — P/N 0891034



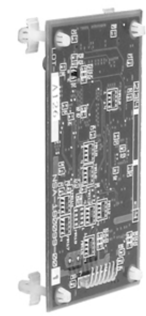
The 4 Port Aspire Mail Plus Expansion Daughter Board (P/N 0891034) adds an additional 4 voice mail ports to Aspire Mail Plus, P/N 0891033, (for a total of 8 ports). There is a maximum 1 daughter board per Aspire Mail PCB, with the daughter board using 4 ports.

8 Port Aspire Mail Plus Expansion Daughter Board (8VMDB) — P/N 0891057



The 8 Port Aspire Mail Plus Expansion Daughter Board (P/N 0891057) adds an additional 8 voice mail ports to Aspire Mail Plus, P/N 0891056, (for a total of 16 ports). There is a maximum 1 daughter board per Aspire Mail PCB, with the daughter board using 8 ports.

4 Port Aspire Mail Expansion Daughter Board (4FMDB) — P/N 0891045



The 4 Port Aspire Mail Expansion Daughter Board (P/N 0891045) adds an additional 4 voice mail ports to Aspire Mail. There is a maximum 1 per Aspire Mail PCB, with the PCB using 4 ports.

Aspire Mail Plus AMIS Networking Activation — P/N 0891059

Use the Aspire Mail Plus AMIS Networking Activation (P/N 0891059) to allow you to network multiple voice mails using AMIS protocol. Each voice mail in the network must have the feature activated.

4 Port IntraMail Compact Flash Card — P/N 0892180

The IntraMail will not work with the Enhanced NTCPU (P/N 0891038). It is only tsupported with the 64-Port NTCPU (P/N 0891002) with basic factory -installed PAL chip or with the Feature Upgrade PAL chip (P/N 0891039).

The IntraMail is a plug-in “in-skin” full-featured, DSP-based integrated Voice Mail with Automated Attendant for the Aspire M (with software 4.93 or higher). This card is installed on the DSP Resource daughter board. The 4 port compact flash card provides:

- 4 Voice Mail ports, 8 hours of message storage, and up to 160 mailboxes.

It requires a DSPDBU Daughter Board P/N 0891003.

The IntraMail Automated Attendant answers incoming calls and routes them quickly and efficiently. Integrated Voice Mail features include Conversation Record, Answering Machine Emulation, and Caller ID with Return Call. Interactive Soft Keys guide the display telephone user through the extensive IntraMail feature set. This feature is available in PCPro software 6.00D or higher.

8 Port IntraMail Compact Flash Card — P/N 0892182

The IntraMail will not work with the Enhanced NTCPU (P/N 0891038). It is only tsupported with the 64-Port NTCPU (P/N 0891002) with basic factory -installed PAL chip or with the Feature Upgrade PAL chip (P/N 0891039).

The IntraMail is a plug-in "in-skin" full-featured, DSP-based integrated Voice Mail with Automated Attendant for the Aspire M (with software 4.93 or higher). This card is installed on the DSP Resource daughter board. The 8 port compact flash card provides:

- 8 Voice Mail ports, 16 hours of message storage, and up to 160 mailboxes.

  It requires a DSPDBU Daughter Board P/N 0891003.

The IntraMail Automated Attendant answers incoming calls and routes them quickly and efficiently. Integrated Voice Mail features include Conversation Record, Answering Machine Emulation, and Caller ID with Return Call. Interactive Soft Keys guide the display telephone user through the extensive IntraMail feature set. This feature is available in PCPro software 6.00D or higher.

IntraMail Specifications		
	0892180	**0892182**
Ports:	4	8
Station Mailboxes:	128	
Routing Mailboxes:	16	
Master Mailboxes:	16 (Only 8 of the 16 Master mailboxes are accessible in Aspire S.)	
Total Mailboxes:	160	
Storage Hours:	8 Hours	16 Hours
Answer Tables:	8	
Dial Action Tables:	16	
Programming Interface:	Aspire M telephone programming or Aspire PCPro software 2.30+.	
Remote Programming:	Access Via HTML-based Aspire WebPro or using customer-provided modems with Aspire PCPro.	
Voice Storage Media:	Flash Card (on IntraMail PCB)	
Languages:	1 (English Mnemonic)	

IP Interface Cards

VoIP PCB Installation Note:
When installing a VoIP PCB, the system automatically assigns trunk ports to match the card's port capacity. For example, a 4VOIPU would take 4 trunk ports, the 4VOIPU with a 4VOIPDB would take 8 trunk ports. Extension ports are not reserved until an IP phone is connected to the system. When the first IP phone is plugged in, the system takes the next four consecutive extension ports available and automatically assigns them as IP ports. The next three IP phones installed will use this group of ports. When the fifth IP phone is connected, the next 4 consecutive extension ports available will be assigned as IP ports.

If the number of trunk ports reserved by the system is a concern, install the trunk cards first, then install the VOIPU PCB. This will allow the trunks to be assigned to the COIU, DIOPU, etc. first. If there are not enough trunk ports available for the VoIP PCB, the system will still recognize the card and allow it to be used for IP phones.

If the PCB is not going to be used for trunks, the logical trunk ports can be set to '0' in **Program 10-03-01 : PCB Setup**, but the physical trunk ports are still assigned to the PCB and cannot be used for any other PCB unless the PCB is deleted from the slot in **Program 90-05 : Slot Control**.

The Aspire VoIP supports H.323, H.325, and H.245 trunks and compressions of G.711, G.723.1, and G.729.

4 CH VoIP Media Gateway (4VOIPU) — P/N 0891042

The 4VoIP PCB is used for converting the RTP (Real Time Transfer Protocol) packets via the IP network and PCM highway. The IP telephones are connected directly to the IP bus. When IP phones need to be connected to a conventional PCM-based digital circuit, this PCB converts the IP packet signal into a PCM signal format and connects to the PCM time division switch.

The 4VOIPU PCB is required in order for IP telephones to communicate with non-VoIP Aspire phones, as well as to place or receive outside calls. This PCB can be used with either the Basic or Enhanced version of the NTCPU (P/N's 0891002 and 0891038).

The 4VOIPU PCB provides:

- 4VOIPU PCB provides up to 4 channels
- Connector for the VOIPDB daughter board (providing an additional 4 channels)
- 1 PCB status LED
- 1 run/block switch

A maximum of 16 PCBs per system are allowed, providing 64 channels (with the daughter board, 128 channels) and it can be plugged into any universal slot. The PCB will consume 4 extension ports and, if available, 4 trunk ports.

If a separate software hub is used (and not the 8SHUBU PCB), it should be a 100Base/full duplex hub. To avoid network problems and to ensure good voice quality, do not use a Repeater Hub/10Base.

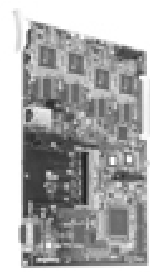
4 CH VoIP Media Gateway Expansion Daughter Board (4VOIPDB) — P/N 0891043

The VOIPDB daughter board provides:

- 4 channels
- Connector for the 4VOIPU PCB (combination provides a maximum of 8 channels per slot)

The VOIPDB is installed on the 4VOIPU PCB with a maximum of 16 daughter boards per system, providing 128 channels (when combining the 4VOIPU and VOIPDB). The daughter board will consume 4 extension ports and, if available, 4 trunk ports. This PCB can be used with either the Basic or Enhanced version of the NTCPU (P/N's 0891002 and 0891038).

16 CH VoIP Media Gateway (16VOIPU) — P/N 0891022



The 16VoIP PCB is used for converting the RTP (Real Time Transfer Protocol) packets via the IP network and PCM highway. The IP telephones are connected directly to the IP bus. When IP phones need to be connected to a conventional PCM-based digital circuit, this PCB converts the IP packet signal into a PCM signal format and connects to the PCM time division switch.

The 16VOIPU PCB is required in order for IP telephones to communicate with non-VoIP Aspire phones, as well as to place or receive outside calls.

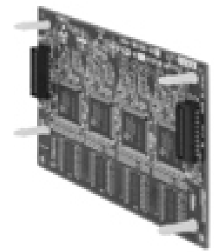
The 16VOIPU PCB provides:

- 16VOIPU PCB provides up to 16 channels
- Connector for the VOIPDB daughter board (providing an additional 16 channels)
- 1 PCB status LED
- 1 run/block switch

A maximum of 16 PCBs per system are allowed, providing 256 channels (with the daughter board, 512 channels) and it can be plugged into any universal slot. The PCB will consume 16 extension ports and, if available, 16 trunk ports.

If a separate software hub is used (and not the 8SHUBU PCB), it should be a 100Base/full duplex hub. To avoid network problems and to ensure good voice quality, do not use a Repeater Hub/10Base.

16 CH VoIP Media Gateway Expansion Daughter Board (16VOIPDB) — P/N 0891023

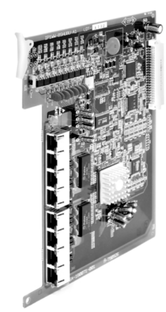


The 16VOIPDB daughter board provides:

- 16 channels
- Connector for the 16VOIPU PCB (combination provides a maximum of 32 channels per slot)

The VOIPDB is installed on the 16VOIPU PCB with a maximum of 16 daughter boards per system, providing 512 channels (when combining the 16VOIPU and VOIPDB). The PCB will consume 16 extension ports and, if available, 16 trunk ports.

10/100BaseT 8-Port Switching Hub (8SHUBU) — P/N 0891021



The LAN PCB is an 8-port switching hub which complies with the ethernet specification for both 100Base-TX and 10Base-T. This PCB is compatible in LAN applications using 10Mbps and 100Mbps. All ports will automatically identify and switch 100Base-TX, 10Base-T and Full/Half Duplex.

The 4VOIPU or 16VOIPU PCB, which is required in order for IP telephones to communicate with non-VoIP Aspire phones, as well as to place or receive outside calls, must be connected to either an external switching hub or to the 8SHUBU PCB.

The PCB plugs into a universal slot, with a maximum of 8 PCBs per system. No ports are consumed with the installation of this PCB. Each PCB provides 8 RJ45 port connectors. These are used to connect to LAN terminals. Depending on the type of LAN terminal, the PCB may not be able to detect the difference between straight cable and cross-cable automatically. If auto-crossover is not functioning, use straight cable for that terminal connection.

IP Station Equipment

34B Aspire IPhone - BK — P/N 0890065

This keyset provides a network connector which allows it to be used with the VoIP feature.

The 34-Button Aspire IPhone has a 3-line, 24-character display with four interactive soft keys for intuitive feature access. In addition, it has 24 user-programmable function keys (with Dual LEDs) for one-button access to co-workers, features and outside lines. The telephone also provides 10 user-programmable One-Touch (Personal Speed Dial) keys and 15 additional fixed feature keys.

The 34-Button Aspire IPhone has a built-in speakerphone and can accept optional IP adapters (PSA, ADA2). You can also assign 110-Button DSS Consoles to these phones, but they must be on site (not networked). It provides Handsfree Answerback, Intercom voice-announcements. In addition, the telephone provides a built-in wall-mount bracket, as well as adjustable legs which allow each phone to be angled at a height which best suits the user.

✍ **At a glance**

34-Button Aspire IP Telephone - Part Number 0890065

Function Keys:	✔	Accepts 110-Button DSS:	✔	Accepts 24-Button DLS:	No
Handsfree (Speakerphone):	✔	Dual LEDs:	✔	ADA Adapter:	No
ADA2 Adapter:	✔	APA Adapter:	No	APR Adapter:	No
CTA Adapter:	No	CTU Adapter:	No	HF-R Adapter	No
ILPA Adapter	✔	:IP Adapter	No	PSA Adapter	✔

4-Button Aspire PoE IP Phone - BK — P/N 0890072B

This 4-button IP keyset is a 802.3af PoE-compliant phone which provides a network connector, allowing it to be used with the VoIP feature.

The 34-Button Aspire IP phone has a 3-line, 24-character display with four interactive soft keys for intuitive feature access. In addition, it has 4 user-programmable function keys (with Dual LEDs) for one-button access to co-workers, features and outside lines. The telephone also provides an 13 additional fixed feature keys.

The 4-Button Aspire IP phone has a built-in speakerphone and provides Handsfree Answerback and Intercom voice-announcements. In addition, the telephone provides a built-in wall-mount bracket.

This phone does **not** provide:

- Data connector for the PC
  Only one ethernet port provided which is used to connect the phone to the system.
- Option Connector
- Headset Connector
- One-Touch Keys
  The One-Touch bins can be accessed, however, through the Soft Keys.
- CONF Key
  Conference can be set up using the Soft Keys or you can define a CONF key on one of the Programmable Function Keys.
- MSG Key
  The Message function can be accessed using the dial pad or through a Programmable Function Key (depending on the action being taken).

2

✍ **At a glance**

4-Button Display Telephone - Part Number 0890072B

Function Keys:	✔	Accepts 110-Button DSS:	✔	Accepts 24-Button DLS:	No
Handsfree (Speakerphone):	✔	Dual LEDs:	✔	ADA Adapter:	No
ADA2 Adapter:	No	APA Adapter:	No	APR Adapter:	No
CTA Adapter:	No	CTU Adapter:	No	HF-R Adapter	No
ILPA Adapter	No	:IP Adapter	No	PSA Adapter	No

34-Button Aspire PoE IPhone - BK — P/N 0890073B

This 34-button IP keyset is a 802.3af PoE-compliant phone which provides a network connector, allowing it to be used with the VoIP feature.

The 34-Button Aspire IPhone has a 3-line, 24-character display with four interactive soft keys for intuitive feature access. In addition, it has 24 user-programmable function keys (with Dual LEDs) for one-button access to co-workers, features and outside lines. The telephone also provides 10 user-programmable One-Touch (Personal Speed Dial) keys and 15 additional fixed feature keys.

The 34-Button Aspire IPhone has a built-in speakerphone and can accept optional IP adapters (PSA, ADA2). You can also assign 110-Button DSS Consoles to these phones, but they must be on site (not networked). It provides Handsfree Answerback, Intercom voice-announcements. In addition, the telephone provides a built-in wall-mount bracket, as well as adjustable legs which allow each phone to be angled at a height which best suits the user.

✍ **At a glance**

34-Button Display Telephone - Part Number 0890073B

Function Keys:	✔	Accepts 110-Button DSS:	✔	Accepts 24-Button DLS:	No
Handsfree (Speakerphone):	✔	Dual LEDs:	✔	ADA Adapter:	No
ADA2 Adapter:	✔	APA Adapter:	No	APR Adapter:	No
CTA Adapter:	No	CTU Adapter:	No	HF-R Adapter	No
ILPA Adapter	✔	:IP Adapter	No	PSA Adapter	✔

H.323 IP Phone — P/N 780005

This UIP300 H.323 IP phone is a business IP phone in an enterprise LAN environment and will be connected to IP PBX systems via an RJ45 network cable.

Standard Telephone Features:
- Alphanumeric LCD display with 2 lines of 24 characters
- 10 LED indicators (Line 1 / Line 2[1] / Status / Mute / Speaker (Headset) / 5 Function keys)
- 12 Key Dial Pad
- 20 Specific Keys (5 Function keys [voice mail or personal speed dial], Menu, Select, Cancel/Del, Transfer, Mute, Redial, Hold, Conference, Speaker, Line 1 and 2 keys, Volume Up and Down keys, Menu Up and Down keys)
- Local Date and Time
- Call Duration Display
- Volume Control for Speaker, Handset, Headset, and Ringer
- Phone Book, Speed Dial, Dial from Call Logs (30 Outgoing Calls, 30 Incoming Calls and 15 Missed Calls)
- Redial, Hold[2], Mute
- Call Waiting, Call Forward, Call Transfer, 3-Way Conference, Do Not Disturb (DND)
- Display Caller ID (Name & Number)
- On-hook Dialing, Handsfree Talking (Full Duplex)
- DTMF Generation
- 8 Ringer Tones

This phone does not support the following:
- The phones cannot send digits after a call has been placed and before it is answered. This means that features which use single digit service codes, such as Voice Over and Barge-In, are not available with this type of phone.
- These phones do not provide P-codes, and therefore, cannot be used with the inDepth application.
- The Message Waiting/Voice Mail LED will not flash when there are new messages.

VoIP Specific Features:
- H.323 v1, 2 Standard Compliant
- Gatekeeper Routed and Direct Routed Call Models
- Voice Codec: G.711 (64kbit/s, u-Law and A-law), G723.1, G729AB
- E.164 Dialing
- Acoustic Echo Cancellation (G.167)
- Rapid Configuration with DHCP or Statically Configured IP Address
- Voice Activity Detection (VAD)
- QoS (IEEE 802.1 p/q Based and DiffServ)
- Jitter Compensation
- 10/100 Base-T Ethernet Interface

[1] Only one phone number will be assigned to this IP phone. Line 2 is not available for a gateway system.

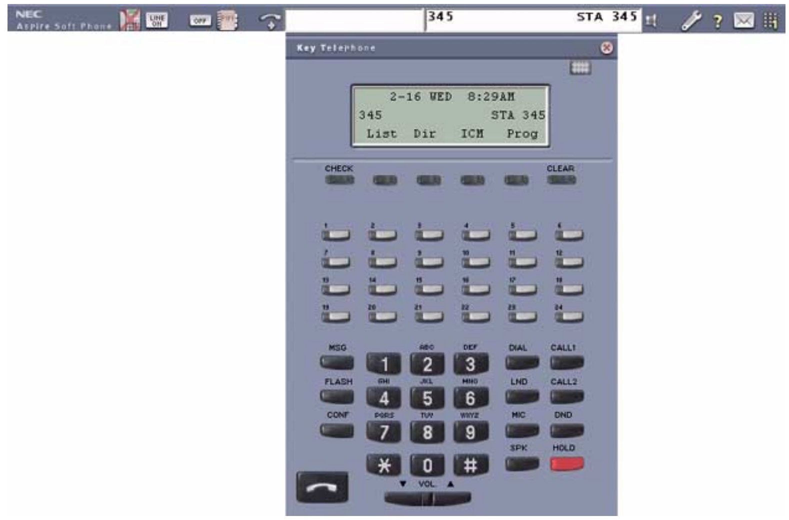
[2] Hold, Transfer, Call Forward and Conference will not be available in the IP address call mode but in the phone number dial mode only.

✍ **At a glance**

H.323 Telephone - Part Number 780005

Function Keys:(not Aspire keys)	✔	Accepts 110-Button DSS:	**No**	Accepts 24-Button DLS:	**No**
Handsfree (Speakerphone):	✔	Dual LEDs:	**No**	ADA Adapter:	**No**
ADA2 Adapter:	**No**	APA Adapter:	**No**	APR Adapter:	**No**
CTA Adapter:	**No**	CTU Adapter:	**No**	HF-R Adapter	**No**
ILPA Adapter	**No**	:IP Adapter	**No**	PSA Adapter	**No**

Aspire Soft Phone — P/N 0893641



Aspire Soft Phone is a business phone which works on a personal computer. It enables various telephony functions using an IP network connecting with the Aspire S/ Aspire system

This allows you to capitalize on the advantages of a converged voice and data network whether you're in the office or on the road. The Aspire Soft Phone application combines traditional business communication needs with the data applications you require.

The Aspire Soft Phone delivers high quality voice via a USB-connected handset or headset or via a PC sound card, microphone and speakers. The Aspire Soft Phone functions include, not only making and receiving calls, but also placing calls on hold, intercom calls, conferencing, etc. The application can display a layout of an Aspire keyset to allow for ease of operation. Using the cursor, simply point and click to operate the Aspire Soft Phone as you would an Aspire keyset.

With Soft Phone software 1.0.3.0 or higher and Aspire system software 4.93 or higher, a USB camera can be connected to the Aspire Soft Phone PCs, allowing a user to make a phone call to another user registered on the same system and provide a video transmission for the call (as long as both users have a USB camera connected and the Aspire Soft Phone application installed with a serial key providing the video function).

In-Line Power Adapter (ILPA-R) — P/N 780122

The In-Line Power Adapter (ILPA-R), which is IEEE 802.3af compliant, detects power from a PoE-compatible ethernet switch and passes it to the IP terminal. The ILPA does the negotiation and detection with the switch and then relays the power to the IP terminal device. This provides an additional way to power the NEC IP terminals (Aspire IPhone or Aspire Keyset with IP Adapter). With this adapter, the IP terminals on the Aspire can be powered using:

- Local power connecting the IP terminal to a local AC wall outlet using the AC-2R Adapter (P/N 780135)
- NEC power supply PoE-managed switch (BlueFire 200/24) (in-line and spare pair detection)
- Aspire 8SHUBU PCB (P/N 0891021) (spare pair detection)
- Cisco Data Switch - CDP supported (in-line and spare pair detection)
- In-Line Power Adapter

Keep the following in mind when installing an ILPA:

- Only IP telephones supported by center feed can be used.
- This adapter can not be used with the H.323 telephones.
- When center feed is used, first unplug the adapter from the ethernet switch before changing the SW1 setting on the back of the adapter.
- Please note that the ILPA-R adapter is intended for use with the Aspire IPhones (P/N 0890065) and IP Adapters (P/N 0890060). Installing any other device into the telephone port of the ILPA-R may result in damage to the device.
- When powering an IP phone using an ILPA-R adapter, the phone should not get connected to a port on the 8SHUBU PCB.

AC Adapter (AC-2R) — P/N 780135

The AC Adapter is required for the IP adapter or Aspire IPhone if external power is needed. Also required for APR, CTU, and Speakerphone optional keyset adapters.

Abbreviated Dialing

Aspire S	Aspire M/L/XL
• Available.	• Available.
• 2000 bins available (0000-1999) for Common and Group Abbreviated Dialing. Up to 8 Abbreviated Dialing Groups available.	• 2000 bins available (0000-1999) for Common and Group Abbreviated Dialing. Up to 64 Abbreviated Dialing Groups available.

Description

Abbreviated Dialing gives an extension user quick access to frequently called numbers. This saves time, for example, when calling a client with whom they deal often. Instead of dialing a long telephone number, the extension user just dials the Abbreviated Dialing code.

There are three types of Abbreviated Dialing: Common, Group and Personal. All co-workers can share the Common Abbreviated Dialing numbers. All co-worker's in the same Abbreviated Dialing Group can share the Group Abbreviated Dialing numbers. Personal Abbreviated Dialing numbers are available only at a user's own extension. To set up Personal Abbreviated Dialing, refer to the "One-Touch Calling" feature. The system has 2000 Abbreviated Dialing bins that you can allocate between Common and Group Abbreviated Dialing. (The Group bins are assigned in groups of 10.)

Each Abbreviated Dialing bin can store a number up to 24 digits long.

When placing an Abbreviated Dialing call, the system normally routes the call through Trunk Group Routing or ARS (whichever is enabled). Or, the user can preselect a specific trunk for the call. In addition, the system can optionally force Common Abbreviated Dialing numbers to route over a specific Trunk Group. User pre-selection always overrides the system routing.

Common Bins Limited to 1000 with Dial Key or #2 Service Code

Though there are 2000 Abbreviated Dialing bins available in the system, once programmed, these bins can currently only be dialed using the Directory Dial feature (Press Directory Dialing Soft Key + ABBc Soft Key + Use arrow keys to locate number or enter the Abbreviated Dial bin name + CALL or SPK to place call.)

The DIAL key and service code #2 operations are not available for any 4-digit Abbreviated Dial common bin number.

DSS Console Chaining

DSS Console chaining allows an extension user with a DSS Console to chain to an Abbreviated Dialing number stored under a DSS Console key. The stored number dials out (chains) to the initial call. This can, for example, simplify dialing when calling a company with an Automated Attendant. You can program the bin for the company number under one DSS Console key (e.g., #200) and the client's extension number under the other (e.g., #201). The DSS Console user presses the first key to call the company, waits for the Automated Attendant to answer, then presses the second key to call the client (extension 400). See Programming below for additional details.

The DSS Console user can also chain to an Abbreviated Dialing number dialed manually, from a Programmable Function Key or a One-Touch Key.

Storing a Flash

To enhance compatibility with connected Centrex and PBX lines, an Abbreviated Dialing bin can have a stored Flash command. For example, storing 9 Flash 926 5400 will cause the system to dial 9, flash the line and then dial 926 5400. The Flash can be stored by the user from their telephone or by the system administrator during system programming.

Using a Programmable Function Key

To streamline frequently-called numbers, an Abbreviated Dialing Programmable Function Key can also store an Abbreviated Dialing bin number. When the extension user presses the key, the phone automatically dials out the stored number. This provides true one-touch calling via a phone's function keys.

Account Codes

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Adjustable Forced Account Code interdigit timer requires software 2.65+.	• Adjustable Forced Account Code interdigit timer requires software 2.65+.

Description

Account Codes are user-dialed codes that help the system administrator categorize and/or restrict trunk calls. The system has three types of Account Codes:

- **Optional Account Codes**
  Optional Account Codes allow a user to enter an Account Code while placing a trunk call or anytime while on a call. This type of Account Code is optional; the system ***does not*** require the user to enter it.
- **Forced Account Codes**
  Forced Account Codes ***require*** an extension user to enter an Account Code every time they place a trunk call. If the user doesn't enter the code, the system prevents the call. As with Optional Account Codes, the extension user can elect to enter an Account Code for an incoming call. However, the system does not require it. ***Forced Account Codes does not block 1-800, 1-888 and emergency assistance (911) calls***.

  Once set up in system programming, you can enable Forced Account Codes on a trunk-by-trunk basis. In addition, Forced Account Codes can apply to all outside calls or just long distance calls. Forced Account Codes for Toll Calls restricts calls according to the following chart:

Number of Digits Dialed	If first digit is not 1	If first digit is 1
1-3	Not allowed	Not allowed
4-7	Allowed - does not require Account Code	Allowed - requires Account Code
More than 7[1]	Allowed - requires Account Code	Allowed - requires Account Code
800 and 888	Allowed - requires Account Code	Allowed - does not require Account Code
011 (International)	Allowed - requires Account Code	N/A
911	Allowed - does not require Account Code	N/A
[1] If you change the local call length in Toll Restriction, this value changes accordingly.		

Timer Modified to Allow for Forced Account Code Interdigit Timer
Programming has been changed which will allow the adjustment of the interdigit timer used for Forced Account Codes. Previously, this was fixed at 3 seconds.

Depending on your software, the system will use the time set in **Program 21-01-04 : System Options for Outgoing Calls - Dial Tone Detection Time** for the interval the system will wait for a user to enter a Forced Account Code. By default, this option is set to 5 (Entries: 0-64800 seconds).

- **Verified Account Codes**
  With Verified Account Codes, the system compares the Account Code the user dials to a list of up to 1000 pre-programmed codes. If the Account Code is in the list, the call goes through. If the code dialed is not in the list, the system prevents the call. Verified Account Codes can be from 3-16 digits long using the characters 0-9 and #. During programming, you can use "wild cards" to streamline entering codes into system memory. For example, the entry 123W lets users dial Verified Account Codes from 1230 through 1239.

Operator Notification
To prevent Account Code abuse, the system can notify the operator each time an Account Code violation occurs. This can happen if the user fails to enter an Account Code (if Forced) or enters a Verified Account Code that is not in the list. The notification is an automatic Intercom call to the attendant and a "*RESTRICT*" message in the operator's display. (If the attendant fails to enter a valid Account Code, the system drops the call.)

Account Codes for Incoming Calls
The system can control the ability of extension users to enter Account Codes for incoming calls. When this option is enabled, a user can dial * while on an incoming call, enter an Account Code, and then dial * to return to their caller. If the option is disabled, any digits the user dials after answering an incoming call outdial on the connected trunk.

Hiding Account Codes
Account Codes can be optionally hidden from a telephone's display. This would prevent, for example, an unauthorized co-worker from obtaining a Verified Account Code by watching the display and making note of the digits that dial out. When hidden, the Account Code digits show as the character "*" on the telephone's display.

Account Code Capacity

Account Codes print along with the other call data on the SMDR record after the call completes. Account Codes can be 1-16 digits in length using 0-9 and #. Verified Account Codes can be from 3-16 digits long.

Redialed Numbers Do Not Contain Account Codes

When using the Last Number Redial, Save or Repeat Dial features, the system will not retain Account Code information. Any number redialed with these features, the user will need to reenter an Account Code.

Note:
If a user enters *12345*203 926 5400*67890*, if the Last Number Redial feature is used, the system dials the number as 203 926 5400*67890*. The *67890* is not treated as an Account Code.

Alarm

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Alarm lets a keyset extension work like an Alarm clock. An extension user can have Alarm remind them of a meeting or an appointment. There are two types of Alarms:

- Alarm 1 (sounds only once at the preset time)
- Alarm 2 (sounds every day at the preset time)

Alarm Reports

Please refer to **Maintenance** (page 152) for information on this feature.

Alphanumeric Display

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Additional language options (entries 10-12 in Program 15-02-01) require software 4.93+.	• Additional language options (entries 10-12 in Program 15-02-01) require software 4.93+.

Description

Multibutton display telephones have a 3-line, 24 character per line alphanumeric display that provides various feature status messages. These messages help the display telephone user process calls, identify callers and customize features.

Analog Communications Interface (ACI)

Aspire S	Aspire M/L/XL
• Available - 8 ACI software ports (4 PGDAD modules max. when used for ACI ports) and 4 ACI Department Groups.	• Available - 96 ACI software ports (48 PGDAD modules max. when used for ACI ports) and 16 ACI Department Groups.

Description

The Analog Communications Interface (ACI) feature uses a PGDAD module to provide two analog ports (with associated relays) for Music on Hold, External Paging or auxiliary devices such as tape recorders and loud bells. The system allows up to 48 PGDAD modules (when used for ACI ports), for a maximum of 96 analog ports. Each PGDAD module requires an unused port on an ESIU PCB.

- **Music on Hold**
  You can connect up to two customer-provided Music on Hold music sources to a PGDAD module. This lets you add additional music sources if the external source on the NTCPU PCB or the internal source are not adequate. By using PGDAD modules, you could even have a different music source for each trunk.

  When the system switches the ACI analog port to a trunk on Hold, the PGDAD relay associated with the ACI analog port closes. You can use this capability to switch on the music source, if desired.

  Extension users can dial the ACI analog port extension number and listen to the connected music source. The PGDAD relay associated with the port closes when the call goes through.

  For Music on Hold, connect the music source to the PGDAD module. Connect the music source control leads to the CTL (control relay) jack. Refer to the Hardware Manual for additional details.

- **External Paging**
  An ACI analog port can also be an External Page output. When connected to customer-provided External Paging equipment, the ACI port provides External Paging independent of the Aspire M/L/XL NTCPU external paging input (the Aspire S CPU does not provide an external paging input). To use the External Paging, an extension user just dials the ACI analog port extension number and makes the announcement. The system broadcasts the announcement from the ACI analog port and simultaneously closes the associated PGDAD relay. You can use the relay closure to control the External Paging amplifier, if required.

  This external paging zone is not included in external all call paging or combination paging (internal and external).

  For External Paging, connect the Paging amplifier to the PGDAD jack. Connect the amplifier control leads to the CTL (control relay) jack. Refer to the Hardware Manual for additional details.

- **Auxiliary Device Control**
  The PGDAD module can control a customer-provided tape recorder. When an extension user dials the ACI analog port extension number, they can automatically start the recorder and activate the record function. When the user hangs up, the recording stops and the tape recorder turns off. For tape recording, connect the tape recorder AUX input jack to the PGDAD jack. Connect the recorder control leads (if available) to the CTL (control relay) jack. Refer to the Hardware Manual for additional details.

  By using Department Calling, you can arrange multiple tape recorders into a pool. When an extension user dials the Department Group pilot number, they reach the first available tape recorder in the pool.

  The relays in the PGDAD module can optionally control customer-provided external ringers (loud bells) and buzzers. When an extension user dials the ACI analog port extension number, the associated PGDAD relay closes and activates the ringer. You could use this capability to control an emergency buzzer for a noisy machine shop floor, for example.

- **ACI Call Recording**
  ACI Call Recording allows you to use a recording device connected to a PGDAD module to manually or automatically record calls. The recording device is typically a customer-provided tape recorder. You can set up ACI Call Recording to output to a single ACI port/recording device or to a pool of ACI ports/devices. With a single device, all calls are stored in a centralized location. With a pool of devices, you'll be sure to have a port available for recording - even in peak traffic periods. You can set up automatic recording on a per trunk or manually on a per extension basis.

  When set up for manual recording, the user presses the ACI Conversation Record key (Service Code 851 + 69 + 0) to begin recording the call from that point. When set up for automatic recording, ACI Call Recording starts automatically as soon as the user places or answers their call. The system can be programed to record all ***incoming*** trunk calls which ring an extension. This includes the following trunk types:
  - Central Office calls programmed to ring the extension.
  - Direct Inward Dialing (DID)
  - Direct Inward Line (DIL)
  - Direct Inward System Access (DISA)
  - Tie lines

  The system can also be programmed to record ***outgoing*** trunk calls, however, this is only possible using E&M tie lines, PRI or BRI trunks.

  ACI Call Recording is not available for intercom calls, transferred calls, or calls placed on hold and answered by an extension with Call Recording enabled. To manually record any type of call (transferred, ICM, outgoing CO trunk, etc.), use the Voice Mail Conversation Record key (Service Code 851 + 78).

Physical Ports and Software Ports

Each PGDAD module consists of a physical port for connection to the phone system and two logical ports. For programming purposes, the ports are also called software ports. The physical port connects to a station position on a EISU PCB. During installation, the first PGDAD module you set up is physical port 1; the second PGDAD module is physical port 2, etc. Each PGDAD module has two software ports, which are numbered independently of the physical ports. Normally, the first PGDAD module set up has software ports 1-2; the second PGDAD module has software ports 3-4, etc. There are a total of 96 software ports (48 PGDAD modules x 2 ports each). During programming, you assign ACI extension numbers and Department Group options to PGDAD software ports, not physical ports. During installation, you connect equipment to the jacks on the PGDAD module that correspond to the software port. Refer to the system Hardware Manual for more installation details.

Aspire Wireless

Aspire S	Aspire M/L/XL
• Not Available.	• Available with software 2.16+ - 120 wireless phones maximum • Program options to adjust transmit and receive level of Aspire Wireless available with 2.63+ software. • Service Codes for registering/unregistering Aspire Wireless phones available with 2.63+ software. • Enhanced Features requires software 4.93+, Aspire Wireless handset version PP4+, and DSIU firmware 1.16+.

Description

The Aspire system provides the ability to use 2.4 GHz Aspire Wireless/DECT (Digital Enhanced Cordless Telecommunication) phones. These phones provide you with the freedom and conveniences of a wireless phone, but in addition, you also have access to features provided by the Aspire system.

The Aspire Wireless (DECT) phone provides additional options as well. Refer to the user guide included with your phone for details on these features.

- 12 Character Alphanumeric Display with Back Light
- LED Indication for Incoming and Unanswered Calls
- Telephone Book with 80 Number Memory Capacity
- Vibration
- Auto Log-In (roaming between different systems)
- 14 Messages Stored
- Stack for 10 Caller ID
- Silent Mode (mute all sounds)
- Redial (last 10 numbers)
- Programming Pause
- Programming of 2 Different Setups (indoor and outdoor)
- Adjustable Volume
- Key Lock
- 9 Different Ring Tones and Adjustable Ring Volume
- Microphone Mute
- Headset Connection
- Automatic Off-Hook (B-Answer)
- R-Key for Transfer and Special Services
- Detects other wireless devices and will temporarily block that frequency range until it becomes available (helps to prevent interference between devices)

Out-of-Range

When a Aspire Wireless phone receives an incoming call and the phone is considered out-of-range (Program 20-22-05 timer expires), what the system does with the call and what the caller hears depends on system programming.

- Extension callers will hear a lock-out tone and may see “Out of Range” on their display.
- DISA callers will following the programming set in Program 25-04-01 : VRS/DISA Transfer Ring Group With No Answer/Busy.
- DID and DIL callers hear ringing and can then be transferred to voice mail or to another extension based on the settings in Program 15-15-05 and 15-15-06.
- Tie line callers will hear the lock-out tone.

The Aspire Wireless requires the following hardware:

- 4-Port Interface PCB (P/N 0891090), 8-Port Interface PCB (P/N 0891091), or 12-Port Interface PCB (P/N 0891092)
- Handset with Battery (P/N 780004)
- Base Station (P/N 780136)
- Charging Cradle (P/N 780137)

Optional Equipment:

- Leather Cover w/Clip for Aspire Wireless (DECT) Handset (P/N 780148)
- Repeater (P/N 780138)
- AC Adapter (P/N 780139)
- Battery (NMh) (P/N 780140)
- Handset Belt Clip (P/N 780141)
- Repeater Programming Kit (P/N 780142)
- Service Tool Kit (P/N 780143)
- Deployment Tool (P/N 780144)
- External Antenna w/Cable for Repeater (P/N 780145)

Prior to deleting an Aspire Wireless (DECT) phone from the system using **Program 91-07-01 : DECT Subscription - Delete**, make sure the DSIU PCB is installed in the system. If the DSIU is removed when Program 91-07-01 is run, the system will retain the DECT setting. This will prevent the Aspire Wireless phone(s) from being registered in the system again.

Programs Available for Gain Adjustment

With software 2.63 or higher, you can adjust the gain of the Aspire Wireless handsets using Program 15-15-07 and 15-15-08.

Register/Unregister Handset by Service Code

The Aspire Wireless telephones can be registered or unregistered at a display keyset using a service code. Note the following conditions when using this feature:

- As this feature uses extension numbers and not port number, it is not possible to have the system select an available port as when using Program 91-06-01.
- This feature only supports the wild card subscription method. It is not possible to specify the IPEI number.
- The wait timer for terminal registration is fixed at 600 seconds.
- This feature can only be used from a display keyset. Digital single line telephones and i-Series phones cannot be used.
- This option is not available for Networked systems.

Enhanced Features Added

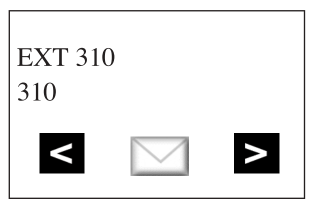
The features for the Aspire Wireless phones have been enhanced. ***Note: In order to enable these enhancements, the Aspire Wireless phones must be updated to software version PP4 and the DSIU PCB must be updated to 1.16 or higher. When both the handset and DSIU are updated to the correct version, Program 90-17-15 will display the version as V01.07.***

- **Calling Party Information with Transfer**
  Previously, when a call is transferred to an Aspire Wireless phone, the Calling Party Number displayed on the LCD was that of the transferring extension - not of the transferred caller. With this release, the Calling Party Number of the transferred caller will now be displayed.

- **Incoming and Outgoing Collision Operation (DSIU firmware changed)**
  When a call collision occurred (an outgoing and incoming call on an Aspire Wireless occurred at the same time), the incoming call would be answered. This is determined by the DSIU PCB firmware. However, previously, the display did not indicate the incoming caller's number.

  With this enhancement, the DSIU PCB firmware has been changed. Now, when a call collision occurs, the phone will not automatically answer the call, but it will suspend the outgoing call operation and indicate the incoming call.

- **Message Waiting Icon Display**
  Previously, the Aspire Wireless phones did not provide an icon display to indicate when a Message Waiting (MW), voice mail message, or general message using the VRS had been left for the Aspire Wireless phone. With this enhancement, when a MW, voice mail message, or general message using the VRS is left, the envelope icon will be shown in the LCD display.

  If the Aspire Wireless is out of range or if the phone is turned off, the message indication will be delayed until the phone returns into range or is powered up.

- **Stand-by Display**
  The Aspire Wireless display has been changed to allow the name and number of the extension to be displayed while in stand-by/idle mode. The stand-by display for the phone is renewed by powering ON/OFF.

  The extension name and number will display even if the Aspire Wireless phone is set with Call Forward.

- **ABB Search**
  An Aspire Wireless user can search the Common and Group Abbreviated Dial numbers for a programmed name by entering one or more characters of the name. Previously, the user needed to know the ABB number to be dialed - it was not possible to search or scroll through the list of numbers programmed in the system.

3

Attendant Call Queuing

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Attendant extensions can have up to 32 incoming calls queued before additional callers hear busy tone. This helps minimize call congestion in systems that use the attendant as the overflow destination for unanswered calls. For example, you can program Direct Inward Lines and Voice Mail calls to route to the attendant when their primary destination is busy. With Attendant Call Queuing, these unanswered calls would normally "stack up" for the attendant until they can be processed.

The 32 call queue total includes Intercom, DISA, DID, DIL, tie line and transferred calls. If the attendant doesn't have an appearance for the queued call, it waits in line on a CALL key. If the attendant has more than 32 calls queued, an extension can Transfer a call to the attendant only if they have Busy Transfer enabled.

Attendant Call Queuing is a permanent, non-programmable system feature.

Automatic Call Distribution (ACD)

Aspire S	Aspire M/L/XL
• Not Available.	• Available with the Basic NTCPU (P/N 0891002) with the PAL upgrade or the Enhanced NTCPU (P/N 0891038) - 64 ACD Groups and 512 ACD Agents.
	• ACD Call Coverage keys supported with 1.06+ software.
	• ACD Queue Status Display supported with 1.11+ software.
	• ACD Hotline Key Shows Agent Status supported with 1.11+ software.
	• ACD MIS software can count overflow calls with 2.63+ software.
	• P command support for AIC log-in requires 4.0E+.
	• Enhanced Overflow options requires 4.94+.
	• Using CVM and Multiple Sources for Announcements requires 5.91+.
	• For more information, refer to the ACD Manual (P/N 0893202).

Description

Automatic Call Distribution (ACD) uniformly distributes calls among member agents of a programmed ACD Group. When a call rings into an ACD Group, the system automatically routes the call to the agent that has been idle the longest. Automatic Call Distribution is much more sophisticated and comprehensive than Department Calling and other group services - it can accurately judge the work load at each agent and distribute calls accordingly. The system allows up to 64 ACD Groups and 512 ACD agents.

You can put any agent in any group. In addition, an agent can be in more than one group. This allows, for example, a Technical Service representation to answer Customer Service calls at lunch time when many of the Customer Service reps are unavailable.

The ACD Master Number is the “extension number” of the whole group. Calls directly ringing or transferred to the ACD Master number enter the group and are routed accordingly. Although the master number can be any valid extension number, you should choose a number that is out of the normal extension range.

Automatic Call Distribution operation is further enhanced by:

- **ACD Call Queuing**
  When all agents in an ACD Group are unavailable, an incoming call will queue and cause the Queue Status Display to occur on the ACD Group Supervisor’s display. The display helps the supervisor keep track of the traffic load within their group. The Queue Status Displays shows:
  - The number of calls queued for an available agent in the group.
  - The trunk that has been waiting the longest, and how long it has been waiting.

  For each ACD Group, you can set the following conditions:
  - The number of trunks that can wait in queue before the Queue Status Display occurs.
  - How often the time in queue portion of the display reoccurs.
  - If the supervisor should hear a Queue Alarm whenever the time in queue portion reoccurs. This alarm is a single beep tone that reminds the supervisor to check the condition of the queue.

- **ACD Overflow (With Announcements)**
  ACD offers extensive overflow options for each ACD Group. For example, a caller ringing in when all agents are unavailable can hear an initial announcement (called the 1st Announcement). This announcement can be a general greeting like, “Thank you for calling. All of our agents are currently busy helping other customers. Please stay on the line and we will help you shortly.” If the caller continues to wait, you can have them hear another announcement (called the 2nd Announcement) such as, “Your business is important to us. Your call will be automatically answered by the first available agent. Please stay on the line.” If all the ACD Group’s agents still are unavailable, the call can automatically overflow to another ACD Group or the Voice Mail Automated Attendant. If all agents in the overflow ACD Group are busy, Lookback Routing automatically ensures that the waiting call will ring into the first agent in either group that becomes free.

  You can assign an ACD Group with any combination of 1st Announcement, 2nd Announcement and overflow method. You can have, for example, a Technical Service group that plays only the 2nd Announcement to callers and then immediately overflows to Voice Mail. At the same time, you can have a Customer Service group that plays both announcements and does not overflow.

  IntraMail does not support ACD Announcement mailboxes.

- **Agent Log In and Log Out Services**
  An ACD Agent can log in and log out of their ACD Group. While logged in, the agent is available to receive ACD Group calls. When logged out, the agent is excluded from the group's calls. The programmable keys and alphanumeric display on an agent's phone show at a glance when they are logged in or logged out.

- **Agent Identity Code (AIC)**
  An Agent Identity Code (AIC) allows ACD agents to log in any extension without setting Program 41-02 (AIC Log In). Using AIC, ACD agents can also log in to multiple ACD groups at the same time (up to 64 ACD Groups). The system will also allow all extensions (up to the system maximum) to log in using the same AIC code. AIC and ACD groups for each work period (mode pattern number) can be set in Program 41-18.

  With older software, P commands are not supported for AIC log on/ log off. Because of this, it is not possible for the inDepth MIS software to determine which ACD groups each extension was a member of or to which ACD groups calls were presented.

  However, with software 4.0E and higher, AIC log on/log off operations are supported. The Aspire P Commands will indicate which ACD Group is being logged onto when an AIC code is entered. Also when an agent logs off, there will be a multiple log of events, one log off event for each previous ACD log on event.

  This feature is requires inDepth software 3.13.0.2b or higher.

- **Multiple Agent Log In**
  ACD agents can log their extension in using multiple AICs (up to 3). The system will also allow all extensions (up to the system maximum) to log in using the same AIC code. For example, even if ACD agent "A" logs in extension 350 with AIC 789, ACD agent "B" can also log in to extension 351 with AIC 789 at the same time.

  In addition to an agent logging in with multiple AIC codes, ACD agents can log in to multiple ACD groups at the same time (up to 64 ACD Groups). AIC and ACD groups for each work period (mode pattern number) can be set in Program 41-18.

  Some conditions with Multiple Agent Log In:
  - ACD agents cannot log in to the system supervisor or group supervisor's extension.
  - In order to log in with AIC, the extension should be set to AIC Log In mode in Program 41-17.
  - If the extension is set to AIC log in mode in Program 41-17, the system will ignore the setting of Program 41-02 for the extension.
  - A supervisor cannot log out an agent logged in by an AIC code.

- **Emergency Call**
  If an ACD Agent needs assistance with a caller, they can place an Emergency Call to their ACD Group Supervisor. Once the supervisor answers the Emergency Call, they automatically monitor both the ACD Agent and the caller. If the agent needs assistance, the supervisor can join in the conversation. Emergency Call can be a big help to inexperienced ACD Agents that need technical advise or assistance with a difficult caller. The supervisor can easily listen to the conversation and then "jump in" if the situation gets out of hand.

- **Enhanced DSS Operation**
  A programmed extension user can use their DSS Console to monitor the status of the ACD Agents within a group. The DSS Console is an essential tool for supervisors. The console key flash rates tell the supervisor at a glance which of the group's agents are:
  - Logged onto the group (i.e., in service)
  - Logged out of the group (i.e., out of service)
  - Busy on a call
  - Placing an Emergency Call to the supervisor
  - Not available or installed

  The ACD Supervisor can also use their console for placing and transferring calls - just like any other extension user.

- **Flexible Time Schedules**
  An ACD Work Schedule lets you divide a day into segments (called Work Periods) for scheduling the activity in your ACD Groups. You can set up four distinct Work Schedules, with up to eight Work Periods in each Work Schedule. Each day of the week has one Work Schedule, but different days can share the same schedule. For example, your Monday through Friday Work Schedule could consist of only two Work Periods. Work Period 1 could be from 8:00 AM to 5:00 PM - when your business is open. Work Period 2 could be from 5:00 PM to 8:00 AM - which covers those times when your business is closed.

- **Headset Operation (With Automatic Answer)**
  An ACD Agent or ACD Group Supervisor can utilize a customer-provided headset in place of the handset. The headset conveniently frees up the user's hands for other work and provides privacy while on the call. In addition, an ACD Agent with a headset can have Automatic Answer. This allows an agent busy on a call to automatically connect to the next waiting call when they hang up.

- **Incoming Call Routing**
  Incoming trunk calls can automatically route to specific ACD Groups. These types of calls ring directly into the ACD Group without being transferred by a co-worker or the Automated Attendant.

- **Rest Mode**
  Rest Mode temporarily logs-out an ACD agent's phone. There are two types of Rest Mode:-
  - **Manual Rest Mode**
    An ACD Agent can enable Manual Rest Mode anytime they want to temporarily log out of the ACD Group. They might want to do this if they go to a meeting or get called away from their work area. While logged out, calls to the ACD Group will not ring the agent's phone.
  - **Automatic Rest Mode**
    When an ACD Group has Automatic Rest Mode, the system will automatically put an agent's phone in Rest Mode if it is not answered. This ensures callers won't have to wait while ACD rings an extension that won't be answered. For keysets, the system enables Automatic Rest Mode for all phones with Rest Mode keys. For SLTs, you must set an option in programming to enable Automatic Rest Mode. If an agent's phone is placed into Rest Mode because a call is not answered, the agent will need to manually cancel Rest Mode in order to log back into the ACD group.

  With a Rest Mode key programmed on an ACD agent's phone, when the agent is in rest mode, the key will be lit. If the Rest Mode key is pressed while an agent is on a call, the key will flash to indicate a pre-Rest Mode status. When the current call is finished, the agent's phone will be in rest mode. The agent can place intercom calls or receive direct incoming calls while in Rest Mode. The ability to receive incoming intercom calls is defined in system programming for each ACD group. Note that an ACD System Supervisor cannot be placed in Rest Mode.

- **Supervisor, ACD Group**
You can designate an extension in an ACD Group to be the group's supervisor. Once assigned as an ACD Group Supervisor, the user can:
- Take the entire ACD Group out of service.
- Check the log out status of each agent after the group taken down.
- Restore the ACD Group to service.

During programming, you can choose one of three modes of operation for each ACD Group supervisor:
- Supervisor's extension cannot receive calls to the ACD Group.
- Supervisor's extension can only receive ACD Group calls during overflow conditions.
- Supervisor's extension receives calls just like any other ACD Group agent (mode 2).

An ACD Group can have only one supervisor. In addition, an extension can be a supervisor for only one ACD Group.

- **Supervisor, ACD System**
You can designate an extension as an ACD System Supervisor. Once assigned as an ACD System Supervisor, the user can:
- Take the all the system's ACD Groups out of service simultaneously
- Check the log out status of each agent after the groups are taken down.
- Restore all the ACD Groups to service simultaneously.

The system can have only one ACD System Supervisor.

- **Work Time**
Work Time temporarily busies-out an ACD agent's phone so they can work at their desk uninterrupted. This gives the agent time to fill out important logs and records as soon as they are finished with their call. There are two types of Work Time:

- **Manual Work Time**
An ACD Agent can enable Manual Work Time any time they need to work at their desk undisturbed. You might prefer this Work Time mode if an agent only occasionally has to fill out follow-up paper work after they complete their call. When the agent is through catching up with their work, they manually return themselves to the ACD Group.

- **Automatic Work Time**
The system implements Automatic Work Time for the agent as soon as they hang up their current call. This is helpful in applications (such as Tech Service groups) where follow-up paperwork is a requirement for every call. When the agent is done with their work, they manually return themselves to the ACD Group.

- **ACD Group Call Coverage Keys**
To help cover calls during peak periods, a keyset can have Call Coverage keys for ACD Groups by assigning the ACD master number to the Call Coverage key. When a call rings into a covered ACD Group, it rings the appropriate ACD Group Call Coverage key, allowing users to pick up incoming ACD calls. The key can ring immediately, after a delay or just flash. The Call Coverage key also facilitates one-button Transfer for an ACD Group. The covering extension does not have to be a member of the ACD Group.

ACD Call Coverage Key LED Pattern	Status
Off	There is no incoming call to the ACD group.
Flashing Red	Incoming call(s) are ringing the ACD group.

- **Hotline Key Shows Agent Status**
  An extension's Hotline keys provide the "normal" Busy Lamp Field (BLF) for co-workers and a unique BLF for ACD Agents. Similarly to the supervisor's DSS Console BLF, the unique BLF shows when the covered agent is in service, out of service or busy on a call. This enhanced BLF gives a department manager, for example, ACD Group monitoring capabilities without having to become a supervisor with a DSS Console.

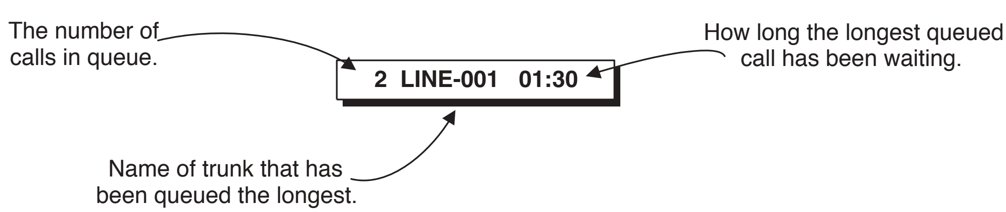
  Hotline gives a keyset user one-button calling and Transfer to another extension (the Hotline partner). Hotline helps co-workers that work closely together. The Hotline partners can call or Transfer calls to each other just by pressing a single key. Enhanced for ACD applications, Hotline provides a unique Busy Lamp Field for ACD agents as well as a BLF for co-workers that are not ACD agents. The charts below show both sets of BLF indications.

BLF For ACD Agents	
When the key is . . .	**The ACD Agent is . . .**
Off	Idle and is not an ACD Agent
On	Busy
Double Wink Off	Making an Emergency Call
Wink Off	Logged off or not installed
Double Wink On	Logged on

BLF For Co-Workers That Are Not ACD Agents	
When the key is . . .	**Your co-worker is . . .**
Off	Idle
On	Busy or ringing
Fast	Flash In Do Not Disturb — All calls (option 3) or Intercom calls (option 2)

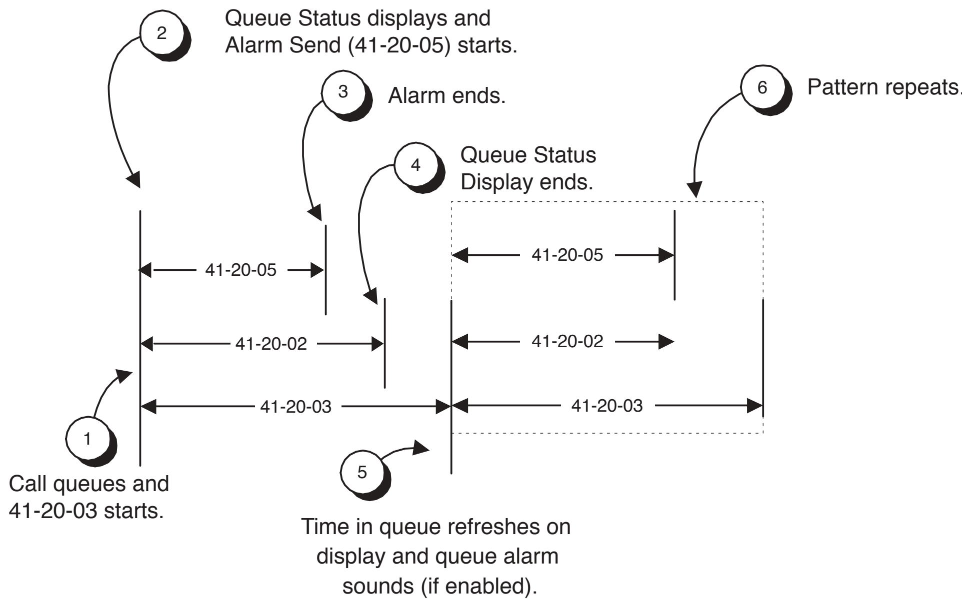
- **Enhanced Supervisor Options**
  An ACD supervisor can individually assign extensions to ACD Groups, and set an agent's status once assigned. This provides the supervisor with tremendous flexibility to reassign agents as work loads vary.

- **Queue Status Display with Scrolling**
  When all agents in an ACD Group are unavailable, an incoming call will queue and cause the Queue Status Display to occur on the ACD Group Supervisor and/or agent's display (based on the Class of Service). The display helps the supervisor keep track of the traffic load within their group. In addition, any display keyset can have a Queue Status Display Check programmable function key. The keyset user can press this key any time while idle, and using the VOL ▲ and VOL ▼, scroll through the Queue Status Displays of all the ACD Groups. The Queue Status Displays shows (see the Queue Status Display illustration below):
  - The number of calls queued for an available agent in the group.
  - The trunk that has been waiting the longest, and how long it has been waiting.



For each ACD Group, you can set the following conditions:

- The number of trunks that can wait in queue before the Queue Status Display occurs.
- How often the time in queue portion of the display reoccurs (see the Queue Status display Timing illustration below).
- Queue Status Display holding time.
- Queue Status Alarm enable/disable.
- Queue Status Alarm sending time.



When Logged Out of ACD Group

When ACD agents are logged out and a call is placed into the ACD queue, the phones of the logged out agents will display the Queue Status and hear the alarm according to the settings defined in system programming. Pressing the Queue Status Display Programmable Function key will return the phone to idle until the timer in Program 41-20-03 expires again.

Notes:

(A.) Do not use both 41-15 and 41-20 to set the ACD queue alarm. Select either one or the other for the system to follow.

Feature	Available in Program 41-15	Available in Program 41-20
Queue Status Display	---	Yes
Queue Status Display Time	---	Yes
Alarm	Yes	Yes
Alarm Send Time	Program 41-15-02 determines the length/interval of the alarm.	Yes
Interval Time of Queue Status Display		Yes
Class of Service	---	Yes

Feature	Available in Program 41-15	Available in Program 41-20
Timing of alarm and display queue status	Alarm triggered after the number of calls in Program 41-15-01 is exceeded.	Alarm triggered after the number of calls in Program 41-20-01 is exceeded. Then follows Program 41-20-03 timing for displaying status.

(B.) If a telephone is not idle, it cannot use the Queue Status Display Programmable Function key.
(C.) The Queue Status Display is not shown and the Queue Alarm is not heard by ACD agents those in Off-Duty mode.
(D.) In order to scroll through the ACD groups queue status, the Queue Status Display Programmable Function key must be used. You cannot scroll when the Queue Status Display appears due to an alarm condition.
(E.) If the Queue Status display and alarm are active and the queued called is answered/ disconnected, the display and alarm will continue until the timers in Program 41-02-02 and 41-20-05 expire.
(F.) When an overflowed call is in queue, the call will be included in its original ACD group's queue and not in the group's queue to which it overflowed.
(G.) The Queue Status is not displayed on a supervisor's phone based on the settings in Programs 41-02-xx. The supervisor must use the Queue Status Display Programmable Function key to view the queue.

3

- **Overflow Announcements from Voice Mail**
  The NVM-Series Voice Mail system can provide the ACD overflow announcements in systems that do not have a DSP daughter board installed for VRS. When a caller queues for an available agent, designated Voice Mail ACD Announcement Mailboxes provide the overflow messages.
  IntraMail does not support ACD Announcement mailboxes.

  - **Escape from Queue with NVM-Series**
    Escape From Queue uses NVM-Series Call Routing Mailboxes for announcement messages to provide callers with enhanced options while in queue. After listening to this type of announcement, they can either wait in queue or dial a digit for an alternate destination. The destination is typically the operator, a mailbox or an extension.

  - **Programmable Wrap-up Timer**
    When an agent finishes their call, the system automatically starts a wrap-up timer and blocks any ACD calls to the agent. This gives them time to complete important logs and records before a new call comes in. When the timer expires, the system returns the agent to the ACD Group to handle new callers.

  - **InDepth and inDepth+**
    InDepth and inDepth+ are Windows-based Management Information Systems that work with the system's built-in ACD. These ACD/MIS systems enhance the system with real time statistics and reports on ACD Group traffic patterns and usage. Refer to the *inDepth and inDepth+* feature for more details.

ACD Group as Overflow Destination

The system can transfer an overflow call to a specific ACD Group or to voice mail using Program 41-09. When Program 41-08-02 : ACD Overflow Destination has the ACD Overflow Destination set to '65', the system will overflow the call to the ACD Group programmed in Program 41-09. (The system will not allow you to program an ACD group with that ACD group as the overflow.) If, while the call is ringing, the extension to which the call was transferred becomes available, both the extension and the overflow ACD group will ring.

Option Available for Counting Overflow Calls in ACD MIS Software

An option is available for use with ACD MIS (such as inDepth/inDepth+) software to count overflow calls. When the number of queued ACD overflow calls exceeds the limit and a busy tone is sent to the caller, the system programming will determine if the call is counted in the ACD MIS software.

Enhancement Provides Additional Overflow Destinations

This feature enhancement provides three new entries for ACD Overflow Transfer destinations. With the older software, only the other ACD group or voice mail could be programmed for the ACD Overflow Transfer destination.

The system can allow either an off-premise number (using a programmed Abbreviated Dial number) or incoming ring group for the ACD Overflow Transfer destination.

Notes:

- When using Off-Premise Overflow transfer and if all trunks are busy, the system will set the ACD Overflow Timer again and wait to retry. When this occurs, the queue information of this call will be set to the oldest queue.
- Once the Overflow call has successfully transferred to the Off-Premise or ring group, the call is removed from the retry queue.
- Once the Overflow call has successfully transferred to the Off-Premise or ring group, the system will notify the ACD-MIS that the call was abandoned by the caller.
- If the user programmed Overflow Transfer to the ring group which is used by the other ACD group, the system will notify ACD-MIS with new call information. Therefore, the ACD-MIS will count the transferred call as new call.

Enhancement Added For Using CVM and Multiple Sources for Announcements

This feature enhances the operation for ACD delay messages. With older software, a CVM (Centralized Voice Mail) could not be used as the source for announcements. In addition, the first and second delay messages were required to use the same source. With software 5.91+, each delay message can separately be assigned to use the ACI, VRS, local voice mail, or centralized voice mail.

The voice mail sources can be from an external voice mail, the Aspire Mail or IntraMail.

Note the following conditions with this feature:

- When using the CVM for announcement message, if the voice mail is located in the local system, the programming should still be entered as for a CVM and not a local voice mail.
- If the message source for the first announcement is not set and all the ACD agents are busy, a caller will continue to hear ring back tone even after the message start time.
- If the message source for the second announcement is not set and all the ACD agents are busy, the following will occur:
  With Program 41-08-08 set to VMI or CVM and Program 41-08-09 set to ACI or VRS
  After the first message is sent, the defined interval tone (RBT, MOH, BGM) for the first announcement message is heard. This tone will continue to be heard until an ACD telephone becomes free.

- If the message source for the second announcement is not set and all the ACD agents are busy, the following will occur:
  With Program 41-08-08 set to VMI and Program 41-08-09 set to CVM
  OR
  Program 41-08-08 set to CVM and Program 41-08-09 set to VMI
  After the 1st message is sent, the second message is sent from the source of the first message.

The table below indicates the required programs when setting Delay Announcements using different sources for the first and second announcements.

2nd Announcement Source	1st Announcement Source			
	ACI	**VRS**	**Local Voice Mail**	**CVM**
ACI	41-10	41-11: 01, 02, 03, 06 22-01-11 41-10: 02, 05	41-19: 01, 02, 03, 06, 08 41-10: 02, 05	41-19: 01, 02, 03, 06, 08 41-10: 02, 05
VRS	41-10: 01, 03 41-11: 04, 05, 06, 07 22-01-11	41-11 22-01-11	41-19: 01, 02, 03, 06, 08 41-11: 04, 05, 06, 07 22-01-11	41-19: 01, 02, 03, 06, 08 41-11: 04, 05, 06, 07 22-01-11
VMI	41-10: 01, 03 41-19: 01, 04, 05, 06, 08	41-11: 01, 02, 03, 06 22-01-11 41-19: 01, 04, 05, 06, 08	41-19	41-19
CVM	41-10: 01, 03 41-19: 01, 04, 05, 06, 07, 08	41-11: 01, 02, 03, 06 22-01-11 41-19: 01, 04, 05, 06, 08	41-19	41-19

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For more on Automatic Call Distribution, refer to the ACD Manual (P/N 0893202).

Automatic Route Selection

Aspire S	Aspire M/L/XL
• Available.	• Available.
• COS option for outgoing calls following access map is available.	• COS option for outgoing calls following access map is available in software 1.11+.
• COS Matching feature is available with 2.63+ software.	• COS Matching feature is available with 2.63+ software.
• Alternate Carrier Access for ISDN trunks requires software 2.63+	• Alternate Carrier Access for ISDN trunks requires software 2.63+.

Description

Automatic Route Selection (ARS) provides call routing and call restriction based on the digits a user dials. ARS gives the system the most cost-effective use of the connected long distance carriers.

ARS is an on-line call routing program that you can customize (like other system options) from a display telephone. ARS accommodates 250 call routing choices - without a custom-ordered rate structure database. With ARS, you can modify the system's routing choices quickly and easily. This is often necessary in today's telecommunications world where the cost structure and service choices frequently change.

The ARS feature can add or delete digits and route calls according to pre-determined levels. When Aspire systems are networked together by a tie line, the networked systems can be called by a system number and a user's extension number, just an extension number, or by using a trunk access code.

ARS Feature Summary

ARS provides:

- **Call Routing**
  ARS can apply up to 24-digit analysis to every number dialed. For programming, ARS provides separate 8-digit and 24-digit tables. Each table can have up to 250 numbers.
- **Dialing Translation (Special Dialing Instructions)**
  ARS can automatically execute stored dialing instructions (called Dial Treatments) when it chooses a route for a call. The system allows up to 15 Dial Treatments. The Dial Treatments can:
  - Automatically insert or delete a leading 1
  - Insert or delete an area code (NPA)
  - Add digits (such as a dial-up OCC number), pauses and waits to the dialing sequence
- **Time of Day Selection**
  For routing purposes, ARS provides ten different day selections (called Time Schedule Patterns). Each Time Schedule Pattern can provide up to 20 time intervals which are assigned to one of the eight day/night modes. The Time Schedule Patterns are then assigned to a day of the week (Monday-Friday, Saturday, Sunday or Holiday).

Basic ARS Operation

When a user places an outside call, ARS analyzes the digits dialed and assigns one of 64 Selection Numbers to the call. The Selection Number chosen depends on which digits the user dialed. ARS then checks the time of day, the day of week and the extension's ARS Class of Service. Based on these call routing options, ARS selects a trunk group for the call and imposes the Dial Treatment instructions (if any).

Class of Service Option Allows Outgoing Calls to Not Follow Access Map

Using this option allows an extension's Class of Service to be set so that ARS does not follow the trunk access map settings (Program 14-07-01 and 15-06-01). The feature allows an extension user to have CO line keys on their phone which allow incoming access only. The user would only have outgoing access on the CO lines when using ARS to place a call.

Class of Service Matching

With the ARS Class of Service Match Access feature, you can determine whether the system should allow a call based on the COS assigned to the Dial Analysis Table (Program 26-02). This change can be used to create a tenant-like application. It will then use the trunk group defined in the Additional Entry in Program 26-02-03 to place the outgoing call.

When this feature is enabled, the calls will be routed in sequential order, and will forward provided the Class of Service for the trunk groups match.

For this feature a new system option, **Program 26-01-06 : Automatic Route Selection Service, COS Match Access**, is added. In adding this feature, the ARS table is expanded from 200 to 400. First 200 entries will be compatible with the previous software as well as the 2.63 software.

Alternate Carrier Access Added for ISDN Trunks
An option is available which allows the system to provide a Transit Network Selection information element for ARS calls using ISDN trunks. This information element identifies a requested transit network. This function is valid only for outbound calls by ISDN trunk.

Local calls do not need Network Selection information since they will be handled by the local exchange carrier (ILEC). If this is the case, ARS is able to distinguish between local and long distance calls and add the Transit Network Selection information element when required.

This option would apply to both PRI and BRI ISDN trunks. If the trunk in the used in **Program 26-02 : Dial Analysis Table for ARS/LCR** is not an ISDN trunk, this code in the dial treatment will be ignored.

Background Music

Aspire S	Aspire M/L/XL
• Available.	• Available.

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Description

Background Music (BGM) sends music from a customer-provided music source to speakers in keysets. If an extension user activates it, BGM plays whenever the user's extension is idle.

Barge In

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Barge In permits an extension user to break into another extension user's established call, including Conference calls. This sets up a Conference-type conversation between the intruding extension and the parties on the initial call. With Barge In, an extension user can get a message through to a busy co-worker right away. If allowed in an extension's Class of Service, multiple users can barge into the same call (up to 32 callers maximum).

There are two Barge In modes: Monitor Mode (Silent Monitor) and Speech Mode. With Monitor Mode, the caller Barging In can listen to another user's conversation but cannot participate. With Speech Mode, the caller Barging In can listen and join another user's conversation.

CAUTION

The use of monitoring, recording, or listening devices to eavesdrop, monitor, retrieve, or record telephone conversation or other sound activities, whether or not contemporaneous with transmission, may be illegal in certain circumstances under federal or state laws. Legal advice should be sought prior to implementing any practice that monitors or records any telephone conversation. Some federal and state laws require some form of notification to all parties to a telephone conversation, such as using a beep tone or other notification methods or requiring the consent of all parties to the telephone conversation, prior to monitoring or recording the telephone conversation. Some of these laws incorporate strict penalties.

Call Coverage

Please refer to the **Multiple Directory Numbers / Call Coverage** (page 162) for information on this feature.

Call Duration Timer

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Program 20-09-06 requires software 2.67+.	• Program 20-09-06 requires software 2.67+.

Description

Call Duration Timer lets a keyset user time their trunk calls on the telephone display. This helps users that must keep track of their time on the phone. For incoming trunk calls, the Call Timer begins as soon as the user answers the call. For outgoing trunk calls, the Call Timer starts about 10 seconds after the user dials the last digit.

Call Forwarding

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Activating Call Forwarding while on a call available with 2.63+ software.	• Activating Call Forwarding while on a call available with 2.63+ software.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.

Description

Call Forwarding permits an extension user to redirect their calls to another extension. Call Forwarding ensures that the user's calls are covered when they are away from their work area. The types of Call Forwarding are:

- **Call Forwarding when Busy or Not Answered**
  Calls to the extension forward when busy or not answered.
- **Call Forwarding Immediate**
  All calls forward immediately to the destination, and only the destination rings.
- **Call Forwarding with Both Ringing**
  All calls forward immediately to the destination, and both the destination and the forwarded extension ring (not for Voice Mail).
- **Call Forwarding when Unanswered**
  Calls forward only if they are unanswered (Ring No Answer).
- **Call Forwarding Follow Me**
  Refer to **Call Forwarding with Follow Me** (page 89) for more.
- **Personal Answering Machine Emulation**
  Allows the extension to emulate an answering machine. Refer to “Voice Mail” for more.

Call Forwarding will reroute calls ringing an extension, including calls transferred from another extension. The extension user must enable Call Forwarding from their phone. To redirect calls while a user is at another phone, use “Call Forwarding with Follow Me”. A periodic VRS announcement may remind users that their calls are forwarded.

Activating Call Forwarding While On a Call

A keyset user can activate or deactivate Call Forwarding while on a call if the phone has a Call Forward Programmable Function Key programmed (15-07-01 or SC: 851 16). Activating Call Forwarding while on a call works only for Call Forward to Station key (Code: 16). Other Call Forward keys (Code: 10-15, 17) do not work while on a call. This option cannot be enabled while the user is hearing a confirmation or warning tone.

With older software, keyset user could only enable Call Forwarding when the phone was idle or when dial tone was heard.

This option is not available for single line telephones.

When Call Forward Follow Me is set up in a chain, forwarding cannot be canceled by the middle keyset while on a call.

Call Forwarding, Centrex

Aspire S	Aspire M/L/XL
• Available with software 5.94+.	• Available with software 5.94+.

Description

With this feature, ***when the system is using all centrex trunks,*** an extension can be forwarded to a destination using a Centrex/PBX line. (*With this feature set, all forwarded calls will have the line flashed, so the system must use only Centrex trunks*.) Forwarding can be set for a keyset, SLT or virtual extension user. The Call Forward can be set manually using a service code (Program 11-11-61) or through system programming (Program 24-08-01, 24-08-02, and 24-08-03). Only Call Forward Immediate, Busy, No Answer, or Busy/No Answer can be used with Centrex lines.

Incoming calls from the COIU PCB must be set to DIL (Program 22-02-01) to the extension or a call can be transferred to the extension. Analog lease lines, analog and digital/PRI DID trunks, and SIP trunks cannot be used with this feature.

Direct Inward Line (DIL)

Trunks which are set for DIL to an extension can follow the Call Forwarding with Centrex setting. With no answer, the timer in Program 24-02-03 is followed to determine when the call should be forwarded. When the timer expires, the system answers the call and sends a hookflash to the trunk. After the timer in Program 21-01-06 expires, the system dials the forwarded number. Then, after the timer in Program 21-01-06 expires again, the trunk goes on hook, completing the forward.

Transfer

If an extension user transfers a trunk call to an extension with Call Forwarding with Centrex set, a hookflash and the destination digits are set to the Centrex trunk. The transferring extension using will hear ringback tone from the CO and must hang up in order to complete the transfer. A call transferred over a Centrex trunk cannot be retrieved. The system will follow the timers in Program 24-02-03 and 21-01-06 for when to transfer and the pause to be inserted before dialing.

When transferring a call to an extension which has set Call Forward with Centrex, forward will not complete until the transferring extension goes on hook. This means that with an Automated Attendant transferring a call, Unscreened Transfer must be used.

When a call is transferred to an extension which has the Call Forwarding with Centrex set, only Immediate and Busy forwarding is available and the transfer must be an unscreened transfer in order to follow the forwarding.

Call Forwarding, Fixed

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.

Description

Fixed Call Forwarding is a type of forwarding that is *permanently* in force at an extension. Calls to an extension with Fixed Call Forwarding enabled automatically reroute - without any user action. Unlike normal Call Forwarding (which is turned on and off by extension users), Fixed Call Forwarding is set by the administrator in system programming. Fixed Call Forwarding complements Voice Mail, for example. The administrator can program Fixed Call Forwarding to send a user's unanswered calls to their Voice Mail mailbox. Each individual user no longer has to manually set this operation.

In system programming, the administrator can set the Fixed Call Forwarding destination and type for each extension and virtual extension. The forwarding destination can be an on- or off-premise extension or Voice Mail. The Fixed Call Forwarding types are:

- Fixed Call Forwarding with Both Ringing (Program 24-06 Option 1)
- Fixed Call Forwarding when Unanswered (Program 24-06 Option 2)
- Fixed Call Forwarding Immediate (Program 24-06 Option 3)
- Fixed Call Forwarding when Busy or Unanswered (Program 24-06 Option 4)
- Fixed Call Forwarding Off-Premise (Program 24-07)

Fixed Call Forwarding reroutes the following types of incoming calls:

- Ringing intercom calls from co-worker's extensions
- Calls routed from the VRS or Voice Mail
- Direct Inward Lines
- DISA, DID and tie line calls to the forwarded extension
- Transferred calls

Fixed Call Forwarding Chaining

Fixed Call Forward Chaining allows Fixed Call Forwards to loop from one extension to the next. For example, you could have the chain 301 ☞ 302 ☞ 303 ☞ 304 set up for Fixed Call Forwarding when Busy. If extension 301 is busy, calls to 301 route to 302. If 302 is also busy, the calls route to 303 and so on. Chaining allows you to set up very basic hunting between co-workers.

Keep the following in mind when setting up Fixed Call Forwarding Chaining:

- If Fixed Call Forwarding Chaining forms a complete Call Forwarding loop (i.e., 301 ☞ 302 ☞ 303 ☞ 301), the system rings the last extension in the chain (303). It does not complete the loop.
- If Fixed Call Forwarding Chaining finds an extension with user-implemented Call Forwarding in the middle of a chain, it rings that extension. It does not continue routing to the other extensions in the chain.

- If one of the extensions in a Fixed Call Forwarding chain has its fixed option set for Both Ringing (1), the system rings that extension. It does not continue routing to the other extensions in the chain.
- The receiving extension's display shows:

STA AAA	AAA is the extension that initially placed the call.
TRANSFER<< STA BBB	BBB is the first extension in the Fixed Call Forwarding chain.

Call Forwarding, Off-Premise

Aspire S	Aspire M/L/XL
• Available.	• Available.
• DSL sets can be used.	• DSL sets can be used.

Description

Off-Premise (OPX) Call Forwarding allows an extension user to forward their calls to an off-site location. By enabling OPX Call Forwarding, the user can stay in touch by having the system forward their calls while they are away from the office. The forwarding destination can be any phone number the user enters, such as a car phone, home office, hotel or meeting room. Off-Premise Call Forwarding can route the off-site phone number over a specific trunk or through a trunk group, Automatic Route Selection or Trunk Group Routing.

Off-Premise Call Forwarding reroutes the following types of incoming calls:

- Ringing intercom calls from co-worker's extensions
- Calls routed from the VRS or Voice Mail [1]
- Direct Inward Lines [1]
- DISA, DID and tie line calls to the forwarded extension [1]
- Transferred calls [1]

OPX Call Forwarding does not reroute Call Coverage keys, Multiple Directory Number keys, or Ring Group calls (i.e., trunk ringing according to Ring Group assignments made in Programs 22-04 and 22-05). Ring Group calls may be forwarded off-premise using voice mail's Call Routing, a dial action table can be created to forward calls to an outside number or Abbreviated Dial number (enter #2001PP with 001 being the # to which the call is forwarded) on timeout using UTRF.

Off-Premise Call Forward for Door Boxes

Off-Premise Call Forwarding allows Door Box callers to be transferred automatically to the pre-programmed external party. The destination telephone number is stored in the Common Abbreviated Dial area. This feature may be used in case a co-worker is out of the office. All incoming calls for their extension will be automatically transferred to their external number (example: cell phone). Off-Premise Call Forward for Door Boxes can be transferred to the external party through ***ISDN lines only***.

1. Off-Premise Call Forwarding can reroute an incoming trunk call only if the outgoing trunk selected has disconnect supervision enabled (see Programming below).

Trunk-to-Trunk Off-Premise Call Forwarding
Use Trunk-to-Trunk Forwarding to automatically forward an incoming trunk call to an outside location. The forwarding destination can be stored in an Abbreviated Dial bin. This feature can be used for trunks which are defined as normal (0) or DID (3) in Program 22-02 : Incoming Call Trunk Setup.

Call Forwarding with Follow Me

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

While at a co-worker's desk, a user can have Call Forwarding with Follow Me redirect their calls to the co-worker's extension. This helps an employee who gets detained at a co-worker's desk longer than expected. To prevent losing important calls, the employee can activate Call Forwarding with Follow Me from the co-worker's phone.

Call Forwarding with Follow Me reroutes calls from the destination extension. To reroute calls from the initiating (forwarding) extension, use Call Forwarding.

Call Forwarding/Do Not Disturb Override

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

An extension user can override Call Forwarding or Do Not Disturb at another extension. This is helpful, for example, to dispatchers and office managers that always need to get through.

Call Pickup Group

Please refer to **Group Call Pickup** (page 133) for information on this feature.

Call Redirect

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Call Redirect allows a keyset user to transfer a call to a pre-defined destination (such as an operator, voice mail, or another extension) without answering the call. This can be useful if you are on a call and another rings in to your extension. By pressing the Call Redirect key, the call is transferred, allowing you to continue with your current call.

This feature works with the following types of calls:

- Normal trunk call
- DID
- DISA
- DIL
- E&M
- ICM

The following types of calls *cannot* be redirected with the feature:

- ACD
- Transferred
- Department Group (all ring mode)
- Door Box
- Virtual Extension

Call Waiting / Camp On

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

With Call Waiting, an extension user may call a busy extension and wait in line (Camp-On) without hanging up. When the user Camps-On, the system signals the busy extension with two beeps indicating the waiting call. The call goes through when the busy extension becomes free. Call Waiting helps busy extension users know when they have additional waiting calls. It also lets callers wait in line for a busy extension without being forgotten.

Callback

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.

Description

When an extension user calls a co-worker that doesn't answer, they can leave a Callback request for a return call. The user does not have to repeatedly call the unanswered extension back, hoping to find it idle.

The system processes Callback requests as follows:

1. Caller at extension A leaves a Callback at extension B.
   Caller can place or answer additional calls in the mean time.
2. When extension B becomes idle, the system rings extension A. This is the Callback ring.
3. Once caller A answers the Callback ring, the system rings (formerly busy) extension B.
   If caller A doesn't answer the Callback ring, the system cancels the Callback.
4. As soon as caller B answers, the system sets up an Intercom call between A and B.

Callback Automatic Answer determines how an extension user answers the Callback ring. When Callback Automatic Answer is enabled, a user answers the Callback ring when they lift the handset. When Callback Automatic Answer is disabled, the user must press the ringing line appearance to answer the Callback ring.

Caller ID

Aspire S	Aspire M/L/XL
• Available - 32 resources available on the CPU for Caller ID (also used for DTMF receivers and Call Progress Tone Detection). The DSPDB will NOT provide any additional resources.	• Available - 32 resources available on the NTCPU for Caller ID (also used for DTMF receivers and Call Progress Tone Detection). The DSPDB provides an additional 32 resources.
• Caller ID information is not available for Aspire Wireless (Aspire Wireless is not supported on Aspire S).	• Caller ID information available for Aspire Wireless with software 1.05+.
• Option for displaying Caller ID name for SLTs is available.	• Option for displaying Caller ID name for SLTs is available with software 1.11+.
• Adding trunk access code to Caller ID is available.	• Adding trunk access code to Caller ID requires software 2.01+.
• Selecting FSK or DTMF for Caller ID requires software 2.63+.	• Selecting FSK or DTMF for SLT Caller ID requires software 2.63+.
• Caller ID Sender Queuing requires software 4.0E+.	• Caller ID Sender Queuing requires software 4.0E+.
• Selecting Caller ID FSK or DTMF signal from an analog trunk requires software 4.0E+.	• Selecting Caller ID FSK or DTMF signal from an analog trunk requires software 4.0E+.
• Caller ID display changed for i-Series phones is not available.	• Caller ID display changed for i-Series phones requires software 4.0E+.
• Temporary storage of Caller ID numbers increased from 16 to 50 with software 4.93+.	• Temporary storage of Caller ID numbers increased from 16 to 50 with software 4.93+.
• Caller ID Detection Time added with software 4.93+.	• Caller ID Detection Time added with software 4.93+.
• Caller ID number deletion enhanced with 4.93+.	• Caller ID number deletion enhanced with 4.93+.
• Flexible Ringing by Caller ID requires software 4.93+.	• Flexible Ringing by Caller ID requires software 4.93+.
• Flexible Ringing by Caller ID on a per-trunk basis requires software 5.91+.	• Flexible Ringing by Caller ID on a per-trunk basis requires software 5.91+.
• Timer for displaying Caller ID record requires software 5.91+.	• Timer for displaying Caller ID record requires software 5.91+.

Description

Caller ID allows a display keyset to show an incoming caller's telephone number (called the Directory Number or DN) and optional name. The Caller ID information is available as pre-answer display. With the pre-answer display, the user previews the caller's number before picking up the ringing line. If Caller ID is enabled in the system, a user can check the Caller ID for a parked call using their Park key and possibly the Flash key (depending on programming). Refer to the table on the following page for the available Caller ID displays.

Second Call Display
While busy on a call, the telephone display can show the identity of an incoming trunk or Intercom call. For incoming trunk calls, the display will show the Caller ID or ANI data or the trunk's name if Caller ID or ANI are not installed. (See the T1 Trunking feature for more on ANI compatibility.) For incoming Intercom calls, the display will show the calling extension's name.

Caller ID supports the telco's Called Number Identification (CNI) and Called Number Delivery (CND) service, when available. These services provide the Caller ID information (i.e., messages) between the first and second ring burst of an incoming call. There are two types of Caller ID message formats currently available: Single Message Format and Multiple Message Format. With Single Message Format, the telco sends only the caller's phone number (DN). The DN is either 7 or 10 digits long. In Multiple Message Format, the telco sends the DN and the caller's name. The DN for this format is also 7 or 10 digits long, and the name provided consists of up to 15 ASCII characters.

Telephone's display can show up to 12 Caller ID digits (for non-ACD calls).

Once installed and programmed, Caller ID is enabled for all types of trunk calls, including:
- Ring Group calls
- Calls transferred from another extension
- Calls transferred from the VRS
- Calls transferred from Voice Mail (unscreened)
- Direct Inward Lines (DILs)

Caller ID temporarily stores 16 or 50 calls (total of abandoned and answered/unanswered), depending on your software version. New calls replace old calls when the buffer fills.

Temporary Memory
An unanswered call will cause the Call History key (PGM 15-07 or SC 851: 08) to flash, indicating a new call has been placed in the temporary memory. If enabled in programming, the telephone's display will show "CHECK LIST".

This Caller ID data from the temporary memory can be saved in either Abbreviated Dial bins or in One-Touch keys making them available for placing future calls.

Caller ID Displays	
Abbreviation	**Description**
Absence code	Absence Reason Code P displays as PRIVATE Absence Reason Code O displays as NO CALLER INFO
CID-num CID-name Trunk name	Caller ID number (Provided by telco) Caller ID name (Provided by telco) Trunk name provided by phone system (Program 14-01-01)
NN:NN:NN HH:MM:SS YY:MM:DD	System's Call Timer display System Time System Date

Conditions	Row	Pre-Answer Display	Post-Answer Display	Display When Reviewing
With Caller ID name and number	1	Trunk name	Trunk name	CID-name
	2	CID-num CID-name	CID-num CID-name	YY:MM:DD HH:MM:SS
With Caller ID number Without Caller ID name With name absence code	1	Trunk name	Trunk name	CID-num
	2	CID-num	CID-num	YY:MM:DD HH:MM:SS
Without Caller ID number With Caller ID name With number absence code	1	Trunk name	Trunk name	CID-name
	2	CID-name	CID-name	YY:MM:DD HH:MM:SS
Without Caller ID number Without Caller ID name With number & name absence codes	1	Trunk name	Trunk name	No Entry
	2	Name Absence Code	Name Absence Code	
Without Caller ID number Without Caller ID name Without number absence code	1	Trunk name	Trunk name	No Entry
	2	Ringing	Ringing	

Outputting Caller ID Data
The system includes the Caller ID data on the SMDR report. The report provides the incoming call's DN in the DIALED NUMBER field. The CLASS field shows PIN (just like all other incoming calls).

Caller ID Digits to Voice Mail
A Caller ID/ANI trunk can send Remote Log-On Protocol with Caller ID digits to the voice mail. When a trunk '001' receives the Caller ID as '12345', the protocol becomes '***60001*12345*'.

Display Reason for No Caller ID Information
With Caller ID enabled, the system will provide information for analog calls that do not detect the Caller ID information. If the Caller ID information is restricted, the telephone display will show "PRIVATE". If the system is not able to provide Caller ID information because telco information is not detected, then the display will show "NO CALLER INFO".

Calling Party Number Information
When using the Aspire Wireless telephone, the system can provide the Caller ID information for an external call if it is provided by the telco.

Option to Enable Caller ID Name for SLT
System programming provides an option for single line telephones to display Caller ID.

Add Trunk Access Code to Caller ID with Aspire Wireless Phones
Aspire Wireless (DECT) phones on the Aspire can hold incoming call history. This history is created based on the Caller ID information element contained in the call's Setup message which is transmitted from the Aspire. This information allows users to return calls dialing the number stored.

The stored number, however, does not contain the trunk access code. Without this code, the system may not be able to seize an outside line to complete the call.

With this feature, when an Aspire Wireless user receives an incoming trunk call, the trunk access code defined in programming can be added to the Caller ID. This will allow the system to seize an outside line and then dial the stored number.

- This function is only applied to incoming analog trunks. It does not apply to incoming extension calls.
- Caller ID must be available for this feature to work.
- Caller ID Edit Mode (Program 20-19-03) must be enabled.
- The maximum number of Caller ID digit is 20. If the total number of digits (trunk access code (Program 10-02-05) and Caller ID) is over 20, the remaining Caller ID digits are not dialed.
  For example:
  Trunk Access Code (Program 10-02-05): 123456#* (8 digits)
  Incoming Caller ID: 12345678901234567890 (20 digits)
  Aspire Wireless Dials: 123456#*123456789012

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Caller ID Sender Queuing Added
The Aspire system can provide Caller ID (calling party number) to a single line telephone which has a display.

With older software, if all Caller ID sender resources were busy in the system, the call would ring the SLT without any Caller ID information displayed. With this enhancement, the system can queue the incoming call to the single line telephone if the system Caller ID sender resources are busy. With this option, Program 20-19-05 is added.

While an incoming call is waiting in queue, if the SLT user lifts their handset, they will hear silence (no dial tone) and can not dial out. When the SLT user goes back on hook, the system immediately sends the queued call to the SLT without Caller ID.

Option Available for FSK or DTMF Type for SLT
An option (Program 15-03-11) is available for the Caller ID which allows you to select either FSK or DTMF as the Caller ID type to be received by a single line telephone.

Option Available for FSK or DTMF Type from Analog Trunk
An option (Program 14-02-16) is available for the Caller ID which allows you to select the type of Caller ID signal from an analog trunk - FSK or DTMF.

Name and Number Limited Display with DSTU PCB
The Caller ID feature has been enhanced to provide better functionality with the i-Series telephones when using a DSTU PCB.

Previously with the i-Series telephones, the Caller ID name was limited to the 10 left most characters and the Caller ID number was limited to 9 digits as both were included on the second line of the display.

With software 4.0E or higher, the name and number will each appear on a separate line, allowing for a maximum of 20 characters/digits per line.

- When there are more than 20 characters set in Program 20-20 : Message Setup for Non-Caller ID Data, either the first or last character(s) will be missing (based on the entry in Program 20-19-01).
- This feature option is available for i-Series telephones only - Aspire telephones will not follow the setting in Program 20-02-15.
- The display of both Caller ID number and name is shown only while the call is ringing.
- The CO name/DID name is not displayed during ringing if Program 20-02-15 is set to "1" or ’2".
- If Program 20-09-06 : Class of Service Options (Incoming Call Service) : Incoming Time Information Display is set to "1" (call time displayed), the first line will display the time and date.

Flexible Ringing by Caller ID
The Flexible Ringing by Caller ID feature provides several different options for rerouting calls based on the Caller ID received.

- **Reject/Reroute "Private" Caller ID Calls**
  When an analog or ISDN trunk call is received with “Private” Caller ID information, the system can reject the call by playing a VRS message or it can route the call to an alternative extension or incoming ring group programmed in Program 22-18-01.

- **Reject/Reroute Based on Entry in ABB Table**
  When an analog, ISDN or IP trunk call is received with regular Caller ID information, the system can reject the call by playing a VRS message if the Caller ID number matches the ABB group number programmed in Program 22-16-01 and ABB entry in Programs 13-02-01 and 13-04-01.

  The analog, ISDN or IP trunk call can also be routed to an alternative extension or incoming ring group if the Caller ID number matches the common ABB table (Program 13-04).

This option can block calls on all trunks or it can be set on a per-trunk basis.

Refer to **Central Office Calls, Answering** (page 97) for the required programming for this feature.

Center Telephone Book

Aspire S	Aspire M/L/XL
• Available - requires software 4.93+.	• Available - requires software 4.93+.

Description

The Center Telephone Book is a new feature providing each user up to two personal telephone books which can be used to store numbers. The system allows multiple extensions to be assigned the same telephone book - this allows users to share commonly used numbers.

- With the Aspire S, up to 50 books of 300 entries can be stored; with the Aspire M/L/XL, up to 100 books of 300 entries can be stored.

- The Telephone Books can be password protected.
- Numbers can be stored using Last Number Dialed and Caller ID

The following key operation can be used with the Telephone Book:

- HOLD button - User can enter the data. (same as the Abbreviated Dial input)
- LND button - User can input the character: @ (same as the Abbreviated Dial input)
- MIC button - User can input the character: P (same as the Abbreviated Dial input)
- FLASH button - User can change the input mode or input the character: R. (same as the Abbreviated Dial input)
- CLEAR button - User can finish the telephone book operation.
- CONF button - User can delete the characters (same as the Abbreviated Dial input) or the user returns back one page.
- VOL UP button - User can change the edit item or page OR the user can start a search.
- VOL DOWN button - User can change the edit item or page OR the user can start a search.
- SOFT KEY button - User can select the function.
- Dial pad - User can input the data or select list.

Central Office Calls, Answering

Aspire S	Aspire M/L/XL
• Available - 8 trunks.	• Available - 200 trunks.
• Additional trunk ring tones available.	• Additional trunk ring tones added with software 1.02+.
• Defining CODEC Filter settings available.	• Defining CODEC Filter settings available with software 1.04+.
• Sidetone Volume Setup available.	• Sidetone Volume Setup added in 1.04+ software.
• Flexible Ringing by Caller ID requires software 4.93+.	• Flexible Ringing by Caller ID requires software 4.93+.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.
• Flexible Ringing by Caller on a per-trunk basis requires software 5.91+.	• Flexible Ringing by Caller on a per-trunk basis requires software 5.91+.

Description

The system provides flexible routing of incoming CO (trunks) calls to meet the exact site requirements. This lets trunk calls ring and be answered at any combination of system extensions. For additional information on making trunk ring, refer to the Ring Group feature.

Delayed Ringing

Extensions in a Ring Group can have delayed ringing for trunks. If the trunk is not answered at its original destination, it rings the DIL No Answer Ring Group (this ring group applies to DIL or non-DIL trunks). This could help a secretary that covers calls for their boss. If the boss doesn't answer the call, it rings the secretary's phone after a programmable interval.

Universal Answer
Universal Answer allows an employee to answer a call by going to any keyset and dialing a unique Universal Answer code. The employee doesn't have to know the trunk number or dial any other codes to pick up the ringing trunk. You'll normally set up Universal Answer along with Universal Night Answer (see "Night Service"). When a Universal Night Answer call rings the External Paging, an employee can answer the call from the first available phone. You might also want to use Universal Answer in a noisy warehouse or machine shop where the volume of normal telephone ringing is not adequate. After hearing the ringing over the Paging, an employee can then easily pick up the call from a shop phone. See "Night Service" for more on Universal Night Answer.

The Automatic Answer of Universal Answer Calls option (Program 20-10-07) determines whether or not the extension has the Auto Answer feature for ringing calls. This option allows a user to simply lift the handset to answer a ringing call; they no longer need to dial the service code.

Display Reason for Transfer
When incoming DID, DISA, DIL or ISDN calls are transferred to another extension or ring group due to a Call Forward or DND setting, the reason for the transfer can be displayed on the phone receiving the transferred call. The extension user can then recognize why they are receiving the call. This feature requires a display telephone in order to view the message.

Clear Down
When on a speakerphone call only, when an outside caller hangs up, the Aspire will return the keyset to an idle condition. The user does not have to press the SPK key to hang up. This feature is automatic and no programming is required.

Additional Trunk Ring Tones
Additional options have been added for the ring tone patters for an incoming call. Previously, in Program 22-03 the system provided ring tone patterns 1-4. With software 1.02 and higher, Melody 1 - Melody 5 are now available as well.

CODEC Filter Data Setup Program Added
When **Program 81-07-01 : CODEC Filter Setup for Analog Trunk Ports** is set to "4 - Specified Data", the system will use the settings in **Program 81-09 : COIU CODEC Filter Data Setup**.

These values should not be changed from their default settings unless directed by NEC'S Technical Service department.

The side tone of the COIU is adjusted using all 16 values, however, special software is required in order to compute these values. The setting is not proportional to the gain level. To change these values, contact NEC's Technical Service department for assistance.

Sidetone Volume Setup
A new option has been added to allow the system programming for the keyset side tone volume. There are two levels, based on whether the connected trunk is a digital trunk or analog trunk.

Flexible Ringing by Caller ID
The Flexible Ringing by Caller ID feature provides several different options for rerouting calls based on the Caller ID received.

- <u>**Reject/Reroute "Private" Caller ID Calls**</u>
  When an analog or ISDN trunk call is received with "Private" Caller ID information, the system can reject the call by playing a VRS message or it can route the call to an alternative extension or incoming ring group programmed in Program 22-18-01.

- **<u>Reject/Reroute Based on Entry in ABB Table</u>**
  When an analog, ISDN or IP trunk call is received with regular Caller ID information, the system can reject the call by playing a VRS message if the Caller ID number matches the ABB group number programmed in Program 22-16-01 and ABB entry in Programs 13-02-01 and 13-04-01.

  The analog, ISDN or IP trunk call can also be routed to an alternative extension or incoming ring group if the Caller ID number matches the common or group ABB table (Program 13-04).

This option can block calls on all trunks or it can be set on a per-trunk basis.

Notes:

- Caller ID Matching
  The system compares the Caller ID and programmed ABB dial and allows/denies as indicated below.

Caller ID	ABB Dial	Result
2039261111	2039261111	Matched
2039261111	20392611119	Matched
2039261111	203	Matched
9261111	2039261111	Unmatched
2039261111	9261111	Unmatched

- The ABB dial table is searched from the starting number and the first match result is used.
- The maximum number of VRS message channels that can be used simultaneously is 16 (Aspire M/L/XL) and 8 (Aspire S).
- With ISDN trunks, the unrestricted digital call can not use this feature.
- This feature does not work with incoming trunk calls via Networking (from the other system). In this case, the refuse/routing program must be programmed in the system that has those trunks. Routing to the other system’s extension is available.
- When Program 13-04 is used, it will override the setting in Program 22-02-01 : Incoming Call Trunk Setup.
- Program 13-04 will follow Common or Group Abbreviated Dial numbers.

Central Office Calls, Placing

Aspire S	Aspire M/L/XL
• Available - 8 trunks.	• Available - 200 trunks.
• Additional trunk ring tones added with 1.02+.	• Additional trunk ring tones added with 1.02+.
• Trunk Port Disable feature available.	• Trunk Port Disable feature available with software 1.00+.
• Defining CODEC Filter settings available.	• Defining CODEC Filter settings available with software 1.04+.
• Sidetone Volume Setup available.	• Sidetone Volume Setup added in 1.04+ software.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.

Description

The system provides flexibility in the way each extension user can place outgoing trunk calls. This lets you customize the call placing options to meet site requirements and each individual's needs. A user can place a call by:

- Pressing Line Keys or "Loop Keys"
- Pressing a Trunk Group (i.e., loop) key
- Pressing a Trunk Group Routing (dial 9) key
- Dialing a code for a specific trunk (#9 + the trunk number)
- Dialing a code for a Trunk Group (804 + group number)
- Dialing a code for Trunk Group Routing or ARS (9)
- Dialing an Alternate Trunk Route Access Code (which you must define)

Trunk Port Disable

The system provides a service code (Default: 145) which can be used by an extension user to block a trunk for outgoing calls. The user which busied out the trunk will still have access to it. All other users will be blocked from seizing it to place an outgoing call. The trunk, however, can still be answered by any users programmed with the trunk access.

Clear Down

When on a speakerphone call only, when an outside caller hangs up, the Aspire will return the keyset to an idle condition. The user does not have to press the SPK key to hang up. This feature is automatic and no programming is required.

Additional Trunk Ring Tones

Additional options have been added for the ring tone patters for an incoming call. Previously, in Program 22-03 the system provided ring tone patterns 1-4. With software 1.02 and higher, Melody 1 - Melody 5 are now available as well.

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These values should not be changed from their default settings unless directed by NEC'S Technical Service department.

The side tone of the COIU is adjusted using all 16 values, however, special software is required in order to compute these values. The setting is not proportional to the gain level. To change these values, contact NEC's Technical Service department for assistance.

Sidetone Volume Setup

A new option has been added to allow the system programming for the keyset side tone volume. There are two levels, based on whether the connected trunk is a digital trunk or analog trunk.

Class of Service

Aspire S	Aspire M/L/XL
• Available - 15 Classes of Service.	• Available - 15 Classes of Service.
• Aspire Wireless registration by service code option not available (Pgm 20-07-21).	• Aspire Wireless registration by service code option available with 2.63+ software (Pgm 20-07-21).
• Incoming Time Information Display (Pgm 20-09-06) requires software 2.67+.	• Incoming Time Information Display (Pgm 20-09-06) requires software 2.67+.
• Items 24 and 25 require software 4.93+ (Pgm 20-07).	• Items 24 and 25 require software 4.93+ (Pgm 20-07).
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.

Description

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Class of Service (COS) sets various features and dialing options (called items) for extensions. The system allows any number of extensions to share the same Class of Service. An extension can have a different Class of Service for each of the Night Service modes. This lets you program a different set of dialing options for daytime operation, nighttime operation and even during lunch breaks. An extension's Class of Service can be changed in system programming or via a Service Code (normally 177).

A Class of Service program has been added to some of the timers which affect extensions and trunks. Each timer Class of Service has 15 different classes (01-15), plus the option to use the system-wide timer (0).

Class of Service Options (Administrator Level), Program 20-07

This option. . .	Is used with . . .	Default	
		COS 01-14	COS 15
Manual Night Service Enabled	Night Service	Disabled	Enabled
Changing the Music on Hold Tone	Music on Hold	Disabled	Enabled
Time Setting	Time and Date	Enabled	Enabled
Storing Abbreviated Dialing Entries	Abbreviated Dialing	Enabled	Enabled
Set/Cancel Automatic trunk-to-Trunk Forwarding	Transfer	Disabled	Disabled
Programmable Function Key Programming (Appearance Level)	Programmable Function Keys	Enabled	Enabled
Forced Trunk Disconnect (analog trunk only)	Forced Trunk Disconnect	Disabled	Enabled
Trunk port disable	Central Office Calls, Placing	Disabled	Enabled

Class of Service Options (Administrator Level), Program 20-07			
This option. . .	Is used with . . .	Default	
		COS 01-14	COS 15
VRS Record	Voice Response System	Disabled	Enabled
VRS General Message Listen	Voice Response System	Disabled	Enabled
VRS General Message Record	Voice Response System	Disabled	Enabled
SMDR printout accumulated extension data	Station Message Detail Recording	Disabled	Enabled
SMDR printout accumulated STG data	Station Message Detail Recording	Disabled	Enabled
SMDR printout accumulated account code data	Station Message Detail Recording	Disabled	Enabled
Aspire Wireless Registration by Service Code	Aspire Wireless	Disabled	Disabled
Set/Cancel Private Call Refuse	Caller ID	Disabled	Disabled
Set/Cancel Caller ID Refuse	Caller ID	Disabled	Disabled
DID Mode Switching	Direct Inward Dialing (DID)	Disabled	Disabled

Class of Service Options (Outgoing Call Service), Program 20-08			
This option . . .	Is used with . . .	Default	
		COS 01-14	COS 15
Intercom Calls	Intercom	Enabled	Enabled
Trunk Calls	Central Office Calls, Placing	Enabled	Enabled
Common Abbreviated Dialing	Abbreviated Dialing	Enabled	Enabled
Group Abbreviated Dialing	Abbreviated Dialing	Enabled	Enabled
Dial Number Preview	Dial Number Preview	Enabled	Enabled
Toll Restriction Override	Toll Restriction Override	Disabled	Disabled
Repeat Redial	Repeat Redial	Enabled	Enabled
Toll Restriction Dial Block	Toll Restriction	Disabled	Disabled
Hotline/Extension Ringdown	Ringdown Extension, Internal/External	Disabled	Disabled
Switching from Handsfree Answerback to Forced Intercom Ringing	Intercom	Enabled	Enabled
Protect for the call mode switching from caller (Internal Call)	Intercom	Disabled	Disabled

Class of Service Options (Outgoing Call Service), Program 20-08			
This option . . .	Is used with . . .	Default	
		COS 01-14	COS 15
Department Group Step Calling	Department Step Calling	Enabled	Enabled
ISDN CLIP	Caller ID	Disabled	Disabled
Call Address Information	Caller ID	Disabled	Disabled
Block Outgoing Caller ID	Caller ID	Disabled	Disabled
Display E911 Dialed Extension Name and Number	E911 Compatibility	Disabled	Disabled
ARS Override of Trunk Access Map	Automatic Route Selection	Disabled	Disabled

Class of Service Options (Outgoing Call Service), Program 20-09			
This option . . .	Is used with . . .	Default	
		COS 01-14	COS 15
Second Call for DID/ DISA/ DIL/ E&M	Central Office Calls Answering	Disabled	Disabled
Caller ID Display	Caller ID	Disabled	Disabled
Sub Address Identification	Caller ID	Disabled	Disabled
Notification for Incoming Call List existence	Caller ID	Disabled	Disabled
Setting Handsfree Answerback or Forced Intercom Ringing	Intercom	Enabled	Enabled
Incoming Time Information Display	Time and Date	Disabled	Disabled

Class of Service Options (Answer Service), Program 20-10			
This option . . .	Is used with . . .	Default	
		COS 01-14	COS 15
Group Call Pickup (Within Group)	Group Call Pickup Call Pickup Group	Enabled	Enabled
Group Call Pickup (Another Group)	Group Call Pickup Call Pickup Group	Enabled	Enabled
Group Call Pickup for Specific Group	Group Call Pickup Call Pickup Group	Enabled	Enabled

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Class of Service Options (Answer Service), Program 20-10			
Group Call Pickup	Group Call Pickup Call Pickup Group	Enabled	Enabled
Directed Call Pickup for Own Group	Directed Call Pickup	Enabled	Enabled
Meet Me Conference and Paging	Meet Me Conference Meet Me Paging	Enabled	Enabled
Automatic Answer of Universal Calls	Universal Answer	Disabled	Disabled
Auto Off-Hook Answer for Call Coverage Keys	Call Coverage	Disabled	Disabled

Class of Service Options (Answer Service), Program 20-11			
This option . . .	**Is used with . . .**	**Default**	
		COS 01-14	**COS 15**
Call Forward Immediately	Call Forwarding	Enabled	Enabled
Call Forward When Busy	Call Forwarding	Enabled	Enabled
Call Forwarding When Unanswered	Call Forwarding	Enabled	Enabled
Call Forwarding (Both Ringing)	Call Forwarding	Enabled	Enabled
Call Forwarding with Follow Me	Call Forwarding	Enabled	Enabled
Unscreened Transfer	Transfer	Enabled	Enabled
Transfer Without Holding	Transfer	Disabled	Disabled
Transfer Information Display	Transfer	Enabled	Enabled
Group Hold Initiate	Hold	Enabled	Enabled
Group Hold Answer	Hold	Enabled	Enabled
Automatic On Hook Transfer	Transfer	Enabled	Enabled
Call Forwarding Off-Premise	Call Forwarding	Disabled	Disabled
Operator Transfer After Hold Callback	Callback	Disabled	Disabled
Trunk to Trunk Transfer Restriction	Transfer	Disabled	Disabled
VRS Personal Greeting	Voice Response System	Enabled	Enabled
Call Redirect	Transfer	Enabled	Enabled
Set/Cancel Department Group Trunk-to-Trunk Forwarding	Tandem Trunking	Enabled	Enabled
No Recall	Transfer	Disabled	Disabled

Class of Service Options (Answer Service), Program 20-11			
This option . . .	Is used with . . .	Default	
		COS 01-14	COS 15
Normal/Extended Park	Park	Disabled	Disabled
Transfer Recall on Originating COS	Transfer	Disabled	Disabled
Restriction for Tandem Trunking on Hang Up	Conference/Tandem Trunking	Disabled	Disabled

Class of Service Options (Supplementary Service), Program 20-13			
This option . . .	Is used with . . .	Default	
		COS 01-14	COS 15
Long Conversation Alarm	Warning Tone for Long Conversation	Disabled	Disabled
Long Conversation Cutoff (Incoming)	Warning Tone for Long Conversation	Disabled	Disabled
Long Conversation Cutoff (Outgoing)	Warning Tone for Long Conversation	Disabled	Disabled
Call Forwarding/DND Override	Call Forwarding/DND Override	Enabled	Enabled
Intercom Off Hook Signaling	Off Hook Signaling	Enabled	Enabled
Automatic Off Hook Signaling	Off Hook Signaling	Enabled	Enabled
Message Waiting	Message Waiting	Enabled	Enabled
Conference	Conference Meet Me Conference	Enabled	Enabled
Privacy Release	Conference	Enabled	Enabled
Barge In Mode	Barge In	Disabled	Disabled
Room Monitor, Initiating Extension	Room Monitor	Disabled	Disabled
Room Monitor, Extension Being Monitored	Room Monitor	Disabled	Disabled
Continued Dialing	Continued Dialing	Enabled	Enabled
Department Calling	Department Calling	Enabled	Enabled
Barge In, Initiate	Barge In	Disabled	Disabled
Barge In, Receive	Barge In	Disabled	Disabled
Barge In Tone/Display	Barge In	Enabled	Enabled
Programmable Function Key Programming (General Level)	Programmable Function Keys	Enabled	Enabled

Class of Service Options (Supplementary Service), Program 20-13			
This option . . .	Is used with . . .	Default	
		COS 01-14	COS 15
Selectable Display Messaging	Selectable Display Messaging	Enabled	Enabled
Account Code/Toll Restriction Operator Alert	Account Codes Toll Restriction	Disabled	Disabled
Extension Name	Name Storing	Enabled	Enabled
Busy Status Display	Intercom	Disabled	Disabled
Display the Reason for Transfer	Transfer	Disabled	Disabled
Privacy Release by Pressing Line Key	Barge In	Disabled	Disabled
Transmission is cut when privacy release is used during trunk to trunk transfer.	Tandem Trunking	Disabled	Disabled
Group Listen	Group Listen	Enabled	Enabled
Busy on seizing virtual extension	Multiple Directory Numbers/ Call Coverage	Enabled	Enabled
Allow COS to be Changed	Class of Service	Disabled	Disabled
Paging Display	Paging, External Paging, Internal	Enabled	Enabled
Background Music	Background Music	Enabled	Enabled
Connected Line identification (COLP)	Caller ID	Disabled	Disabled
Deny Multiple Barge Ins	Barge In	Disabled	Disabled
ACD Supervisor’s Position Enhancement	Automatic Call Distribution	Disabled	Disabled
Block Manual Off Hook Signaling	Off Hook Signaling	Disabled	Disabled
Block Camp On	Call Waiting / Camp On	Disabled	Disabled
Call Duration Timer	Call Timer	Disabled	Disabled
Headset Ringing	Headset Operation	Disabled	Disabled
ACD Queue Display	Automatic Call Distribution	Disabled	Disabled
Do Not Disturb	Do Not Disturb	Enabled	Enabled

Class of Service Options (DISA/E&M Service), Program 20-14			
This option . . .	**Is used with . . .**	**Default**	
		COS 01-14	**COS 15**
First Digit Absorption	Tie Lines	Disabled	Disabled
Trunk Group Routing/ARS Access	Trunk Group Routing Automatic Route Selection	Disabled	Disabled
Trunk Group Access	Trunk Groups	Disabled	Disabled
Common Abbreviated Dialing	Abbreviated Dialing	Disabled	Disabled
Operator Calling	Direct Inward System Access (DISA) Tie Lines	Disabled	Disabled
Internal Paging	Paging, Internal	Disabled	Disabled
External Paging	Paging, External	Disabled	Disabled
Direct Trunk Access	Central Office Calls, Placing	Disabled	Disabled
Forced Trunk Disconnect <Not for ISDN T-point>	Forced Trunk Disconnect	Disabled	Disabled
Call Forward Setting by Remote Via DISA	Call Forwarding Direct Inward System Access (DISA)	Disabled	Disabled
DISA/Tie Trunk Barge In	Barge In	Disabled	Disabled

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Class of Service Timer Data, Program 20-31				
Item	**Default: Class 0 Follows Program**	**Type**	**Timer**	**Default**
01	20-01-08	Extension Timer Class of Service	Trunk Queuing Callback Duration Time	15
02	20-01-09	Extension Timer Class of Service	Callback / Trunk Queuing Cancel Time	64800
03	20-04-03	Extension Timer Class of Service	Call Coverage Delay Interval Time (Virtual Extension Key)	10
04	21-01-02	Extension Timer Class of Service and Trunk Timer Class of Service	Intercom Interdigit Time	10
05	21-01-03	Extension Timer Class of Service and Trunk Timer Class of Service	Trunk Interdigit Time	5
06	21-01-09	Extension Timer Class of Service	Hotline Time Start Time	5
07	22-01-03	Trunk Timer Class of Service	Ring No Answer Alarm Time	60

Class of Service Timer Data, Program 20-31				
Item	Default: Class 0 Follows Program	Type	Timer	Default
08	22-01-04	Trunk Timer Class of Service	DIL/Incoming Ring Group No Answer Time	0
09	22-01-06	Trunk Timer Class of Service	DID Ring-No-Answer Time	20
10	24-01-01	Extension Timer Class of Service and Extension’s Class of Service	Hold Recall Time (Non exclusive Hold)	90
11	24-01-02	Extension’s Class of Service	Hold Recall CallBack Time (Non exclusive Hold)	30
12	24-01-03	Extension Timer Class of Service and Extension’s Class of Service	Exclusive Hold Recall Time	90
13	24-01-04	Extension’s Class of Service	Exclusive Hold Recall Callback Time	30
14	24-01-06	Extension Timer Class of Service and Extension’s Class of Service	Park Hold Time – Normal	90
15	24-02-03	Extension Timer Class of Service	Delayed Call Forwarding Time	10
16	24-02-04	Extension Timer Class of Service and Extension’s Class of Service which performed the blind transfer	Transfer Recall Time	30
17	25-07-02	Trunk Timer Class of Service	DID/DISA No Answer Time (Disconnect or IRG or VM)	30
18	25-07-03	Trunk Timer Class of Service	Disconnect after Re-transfer to IRG	60
19	25-07-07	Trunk Timer Class of Service	Long Conversation Warning Tone Time (Trunk to Trunk)	180
20	25-07-08	Trunk Timer Class of Service	Long Conversation Disconnect (Trunk to Trunk)	10
21	25-07-09	Trunk Timer Class of Service	DISA Internal Paging Time	30
22	25-07-10	Trunk Timer Class of Service	DISA External Paging Time	30
23	31-01-02	Extension Timer Class of Service and Trunk Timer Class of Service	Page Announcement Duration	1200

Computer Telephony Integration (CTI) Applications

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Computer Telephony Integration (CTI) applications automate your office with TAPI compatibility and external PC control. CTI puts your telephone system on the cutting edge of modern office productivity with:

Personal Computer Interface (PCI) / PC Dialing

Use a CTA or CTU Adapter installed in your keyset as a Personal Computer Interface. Installing the TAPI software driver and TAPI compatible software in your personal computer will allow your PC to operate your telephone. The TAPI software driver provides all TAPI Basic Services and a host of TAPI Supplemental Services. See "TAPI Compatibility" for more.

Database Lookup

Provided through Caller ID and TAPI Compatible third-party software (such as ACT!), Database Lookup displays your caller's account information before their call is even answered.

Conference

Aspire S	Aspire M/L/XL
• NTCPU provides 32 conference circuits, allowing each block to have any number conferences with any number of internal or external parties conferenced as long as the total number of conference channels used does not exceed the limit of 32.	• NTCPU provides 2 blocks of 32 conference circuits, allowing each block to have any number conferences with any number of internal or external parties conferenced as long as the total number of conference channels used does not exceed the block's limit of 32.
• Automatic conference on Hang Up available with software 2.10+.	• Automatic conference on Hang Up available with software 2.10+.
• Transfer Call into Conference is available.	• Transfer Call into Conference requires software 1.11+.
• Changing CONF to a Transfer key is available.	• Changing CONF to a Transfer key requires software 1.12+.
• Conference feature enhancements require software 2.63+.	• Conference feature enhancements require software 2.63+.

Description

Conference lets an extension user add additional inside and outside callers to their conversation. With Conference, a user may set up a multiple-party telephone meeting without leaving the office. The Aspire M/L/XL NTCPU provides 2 blocks of 32 Conference circuits - the Aspire S provides 32 circuits, allowing each block to have any number of internal or external parties conferenced up to the block's limit of 32. This means that one extension can Conference up to 31 internal and/or external parties together (the originator would be the 32nd party reaching the maximum of 32). While this

Conference call is active, another user can use the second block of Conference circuits to make the same type of call.

Each block of Conference circuits can have multiple Conference calls, providing there are Conference circuits available. It is not restricted to one Conference per block.

Split (From Conference)
Split allows a user to alternate (i.e., switch) between their callers in Conference. This will allow a dispatcher, for example, to control a telephone meeting between themselves, a customer and a service technician. The dispatcher can meet together with all parties, privately set up a service strategy with the technician and then meet again to set the schedule.

Split cycles through the Conference in the same order in which the Conference was initially set up. If a user places an outside call, conferences extension 302 followed by extension 303, Split will cycle from the trunk, to 302 and finally to 303. The Split cycle then repeats.

Barge Into Conference
If a user's extension has Barge In capability enabled, they can also Barge In on an established Conference. This permits, for example, an attendant or supervisor to join a Conference in an emergency. It also allows a co-worker to leave a conference -- and then rejoin the telephone meeting when it is convenient to do so.

Automatic Conference on Hang Up
A Class of Service option is available, depending on your software version, which will allow or deny an extension user from automatically setting up a Conference/Tandem Trunking call upon hanging up the phone.

Transfer Call Into Conference
An extension with Barge In capability can Transfer a call into an existing Conference. This would allow, for example, an attendant to locate co-workers and then Transfer them into an existing telephone meeting. There is no need for the attendant to locate all the parties at the same time and sequentially add them into the Conference.

Change CONF to a Transfer Key
An option is available which allows an extension's CONF key to be programmed for Conference or for Transfer. When set for Transfer, the user places a call on hold, dials the extension to which it should be transferred, the user presses the CONF key. The call is then transferred. When set for Conference, with an active call, the user presses the CONF key, places a second call, then presses the CONF key twice. All the calls are then connected.

Conference Feature Enhanced
The Conference feature has been enhanced with the ability to:

- Allow a user in an active Conference call to add another party.

  Older software would not allow a call to be added to an existing Conference unless the call was put on hold or park and the Conference reestablished, then adding all parties into the Conference at once.

- Break up a Conference call by pressing Hold.

  With older software, using line keys, if you set up a 3-party (1 internal user and 2 trunks) Conference, if you placed the Conference on hold and pressed one of the line keys, the conference would restart. You could not undo a Conference once it was set by simply placing Hold.

- Drop a trunk or trunks from the Conference.

  While in a Conference, with older software, you could not drop one of the parties.

Conference, Remote

Aspire S	Aspire M/L/XL
• NTCPU provides 32 conference circuits, allowing each block to have any number conferences with any number of internal or external parties conferenced as long as the total number of conference channels used does not exceed the limit of 32.	• NTCPU provides 2 blocks of 32 conference circuits, allowing each block to have any number conferences with any number of internal or external parties conferenced as long as the total number of conference channels used does not exceed the block's limit of 32.

Description

The Remote Conference feature enhances the Conference feature by allowing outside parties to dial a Remote Conference pilot number and a Conference Group number to connect to a Conference call.

The conference circuits on the NTCPU are used to join each party to the conference. A maximum of 32 conference participants is possible for one Conference. With the Aspire M/L/XL system, however, the conference call cannot be split over the NTCPU's conference blocks. This could limit the number of participants if other conference circuits are in use.

With the Aspire M/L/XL, a maximum of 4 simultaneous Remote Conference calls is possible. With the Aspire S, 1 Remote Conference call is possible.

The conference call is password protected so that any user joining the conference would be required to enter a password before being connected.

Conference, Voice Call/Privacy Release

Aspire S	Aspire M/L/XL
• NTCPU provides 32 conference circuits, allowing each block to have any number conferences with any number of internal or external parties conferenced as long as the total number of conference channels used does not exceed the limit of 32.	• NTCPU provides 2 blocks of 32 conference circuits, allowing each block to have any number conferences with any number of internal or external parties conferenced as long as the total number of conference channels used does not exceed the block's limit of 32.

Description

Voice Call Conference lets extension user's in the same work area join in a trunk Conference. To initiate a Voice Call Conference, an extension user just presses the Voice Call Conference key and tells their co-workers to join the call. The system releases the privacy on the trunk, and other users can just press the trunk's line key to join the call.

Voice Call Conference does not use the telephone system features to announce the call. The person initiating the Voice Call Conference just announces it "through the air."

Privacy Mode Toggle Option

The Privacy Mode Toggle option allows an extension user to quickly change an outside call from the non-private mode to the private mode. This would help a workgroup supervisor, for example, that needed to quickly monitor any group member's call. If the supervisor wanted to make a "secure" call, however, they could quickly switch the line's mode and be assured that their call would not be monitored. If the outside call is on a line key, the user just presses the line key to switch modes. If the call is on a loop key, the user presses their Privacy Release function key instead.

For systems using the Privacy Mode Toggle option, trunks initially have the privacy released. If privacy is desired for a trunk, use the toggle option or press the Privacy Release function key to switch modes.

Continued Dialing

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Continued Dialing allows an extension user to dial a call, wait for the called party to answer and then dial additional digits. This helps users that need services like Voice Mail, automatic banking and Other Common Carriers (OCCs).

There are two types of Continued Dialing:

- **Continued Dialing for Intercom Calls**
  Depending on an extension's Class of Service, a keyset user may be able to dial additional digits after their Intercom call connects. In systems with Voice Mail, for example, Continued Dialing lets extension users dial the different options after the Voice Mail answers. Without Continued Dialing, extension users cannot access these Voice Mail options.

- **Continued Dialing for Trunk Calls**
  Continued Dialing gives a user access to outside services like automatic banking, an outside Automated Attendant, bulletin boards and Other Common Carriers (OCCs). After the outside service answers, the user can dial digits for whatever options the services allow. Without Continued Dialing, the system's Toll Restriction will cut off the call after a specific number of dialed digits. See Programming below for additional information.

NOTICE
Continued Dialing may make the system more susceptible to toll fraud.

Cordless II/Cordless Lite II

Aspire S	Aspire M/L/XL
• Available.	• Available with software 1.06+.
• Cordless phones do not display E911 Alarm with software 4.0i or higher.	• Cordless phones do not display E911 Alarm with software 4.0i or higher.
• The ability to press TALK after placing a CO call on Hold without immediately recalling requires software 4.0W+.	• The ability to press TALK after placing a CO call on Hold without immediately recalling requires software 4.0W+.

Description

The Aspire System supports a Cordless Telephone. The DTR-4R-1 Cordless II (P/N 730088) is a 900 MHz spread-spectrum digital cordless telephone that provides mobility, flexibility and convenience for those who spend much of the workday away from their desk. Fully integrated with the telephone system, the DTR-4R-1 Cordless II offers many standard features such as Park, Do Not Disturb, Hotline, Voice Over and Voice Mail. Normally paired with a companion keyset for improved 1-button call coverage capabilities, the DTR-4R-1 Cordless II will also work as a stand-alone telephone.

Complemented by 4 fully programmable function keys (with LEDs), the DTR-4R-1 Cordless II achieves a whole new level of convenience and mobility. An easy-to-read LCD display, volume controls, a rechargeable nickel-cadmium battery pack and a handy belt clip round out the elegant and affordable DTR-4R-1 Cordless II Phone. This phone provides 10 channels.

A second Cordless Phone is also available: the Cordless Lite II (P/N 730087). The Cordless Lite II offers the same features as the Cordless II except that it uses a NiMH battery and has FM modulation (single channel) instead of the spread spectrum modulation. This phone provides 30 channels.

The base unit provides a “Cordless” and “Desk” button which allows the one extension port to switch from the Cordless to an attached Aspire keyset. The phones must be in an idle state when the button is pressed, otherwise any active call will be disconnected.

E911 Alarms

Prior to 4.0i software, if the system was set to display an E911 alarm to telephones (Program 20-08-16), the cordless phones would also indicate the display. There was no way for the users to clear the display of this alarm except by resetting the system. The system has now been changed to exclude the Cordless II and Cordless Lite II phones from displaying the alarm.

Privacy

Cordless telephones are radio devices. Communications between the handset and base of the cordless telephone are accomplished by means of radio waves which are broadcast over the open airways. Because of the inherent physical properties of radio waves, communication can be received by radio receiving devices other than your own telephone unit, consequently, any communications using the cordless telephone may not be private.

Radio Interference

Radio interference may occasionally cause buzzing and humming in your cordless handset or clicking noises in the base unit. This interference is caused by external sources such as TV, fluorescent lighting, or electrical storms. Your unit is not defective. If these noises continue and are too distracting, check around the office to see what appliances may be causing the problem, In addition, it is recommended that the base not be plugged into a circuit that also powers a major appliance because of the potential of interference. For best performance, ensure that the antenna on the base unit is fully extended.

In the unlikely event that other voices or distracting transmissions are heard on the phone, radio signals for another cordless telephone or other source of interference may be the problem. If this type of interference cannot be eliminated, change the telephone to a different channel.

If should also be noted that some cordless telephones operate at frequencies that may cause interference to nearby TVs and VCRs. To minimize or prevent such interference, the base unit should not be placed near or on top of a TV or VCR. If interference is experienced, moving the cordless telephone farther away from the TV or VCR will often reduce or eliminate the interference.

Multiple Cordless Telephones

If you want to use more than one cordless telephone in the same office, they must operate on different channels. Press the channel key to select a channel that provides the clearest communication.

In ideal conditions, multiple spread spectrum type cordless phones (Cordless II) can be utilized in the same environment. However, due to the possible interference problems caused by the bases being placed in close proximity to each other, the following is recommended:

Spread Spectrum Phones (Cordless II P/N 730088)
Where users require greater range on the cordless phones and 3 or less cordless phones are being used at a specific site, we recommend using the spread spectrum cordless phone.

FM Modulation Phones (Cordless Lite II P/N 730087)
Where more than 3 cordless phones are to be used at one specific site, we recommend using the FM modulation cordless phones which have 30-channel capability.

* Note:

1. The range of the phones depends largely on the environmental factors, such as the building structure, the size of the room, RF interference and other electronic equipment installed in the same area. For optimum range and performance, the following is suggested:
2. Place the base units at least 15 feet apart. The performance of the phones become more stable when the distance between the bases is greater.
3. Place the base unit in the center of the coverage area. If the phone will also be used in an outdoor area, like a parking lot, install the base unit in an area close to the window.
4. If a phone experiences interference and noise, press the channel key to select another channel.

Hold on CO Calls No Longer Immediately Recalls

The operation of the Cordless II and Cordless Lite II phones have been enhanced when placing an outside call on hold.

With older software, if an outside caller was placed on hold by a Cordless II/Cordless Lite II user and the user then pressed the TALK button to hang up, the call would immediately recall the user. With software 4.0V+, if the phone has a line key (Program 15-07-01 or SC 851: *01) or a loop key (Program 15-07-01 or SC 851: *02 or *05) programmed, the call will remain on hold and recall only after the hold recall timer has expired (Program 24-01-01). However, without these keys (with software 4.0V only), when placing a call on hold and pressing the TALK button to hang up, the call would immediately ring back.

Software 4.0W further enhances the operation so that a user without a line or loop key programmed can place a caller on hold and press the TALK button. The call will remain on hold and recall only after the hold recall timer has expired (Program 24-01-01).

Intercom calls will still recall immediately if the Cordless II/Cordless Lite II user presses TALK after placing an internal call on hold.

Department Calling

Aspire S	Aspire M/L/XL
• Available - 8 Department Groups.	• Available - 64 Department Groups.

Description

With Department Calling, an extension user can call an idle extension within a preprogrammed Department Group by dialing the group's pilot number. The call would ring the first available extension in the group. For example, this would let a caller dial the Sales department just by knowing the Sales department's pilot number. The caller would not have to know any of the Sales department's extension numbers.

There are two types of routing available with Department Calling: Priority Routing and Circular Routing. With Priority Routing, an incoming call routes to the highest priority extensions first. Lower priority extensions ring only if all higher priority extensions are busy. With Circular Routing, each call rings a new extension (with circular routing, a new call will ring the extension which has been idle the longest in the group). The ringing order is based on an extension's priority. In a Department Group with extensions 310 (Priority 1), 311 (Priority 2) and 312 (Priority 3)

- The first call rings 310.
- The second call rings 311.
- The third call rings 312.
- The fourth call rings 310 and the cycle repeats.

Note: When programming, the high priority extensions have low priority numbers. For example, priority 1 has a higher priority than priority 10.

Overflow Routing

Department Calling also provides overflow routing for extensions within the group. If a user directly dials a busy extension within a Department Group, the system can optionally route the call to the first available group member. The system follows Program 22-15-03 and 22-15-05 timers for playing the periodic VRS message.

User Log Out/Log In

An extension user can log out and log in to a Department Calling Group. By logging out, the user removes their extension from the group. Once logged out, Department Calling bypasses their extension. When they log back in, Department Calling routes to their extension normally. All users can dial a code to log in or log out of their Department Calling Group. A keyset can optionally have a function key programmed for one-button log in and log out operation.

Enhanced Hunting

Department Calling is enhanced with expanded hunting capabilities. Hunting sets the conditions under which calls to a Department Group pilot number will cycle through the members of the group. The hunting choices are:

- **Busy**
  A call to the pilot number will hunt past a busy group member to the first available extension.
- **Not answered**
  A call to the pilot number will cycle through the idle members of a Department Calling group. The call will continue to cycle until it is answered or the calling party hangs up. If the Department Group has Priority Routing enabled, and the highest priority member is busy, the call will not route.
- **Busy or not answered**
  A call to the pilot number will cycle through the idle members of a Department Calling group. The call will continue to cycle until it is answered or the calling party hangs up. Calls into groups with Priority Routing and Circular Routing route identically.
- **Simultaneous ringing**
  All idle members of the Department Group ring simultaneously. Calls do not cycle between group members.

If all members of the Department Group are busy, an incoming or transferred call to the group's pilot number will queue for an available member. Each group has a queue that can hold any number of waiting calls. If a display phone is waiting in queue, the user will see: *WAITING (group name)*. If a transferred call in queue is an outside call, and the system has DSP daughter board installed with the VRS, the queued caller will hear, *"Please hold on. All lines are busy. Your call will be answered when a line becomes free."*

The VRS can also transfer calls to Department Groups. Refer to "Voice Response System (VRS)" feature for more information on setting up the VRS.

The system prevents hunting to a Department Group extension if it is:

- Busy on a call
- In Do Not Disturb
- Call Forwarded

Department Step Calling

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

After calling a busy Department Calling Group member, an extension user can have Department Step Calling quickly call another member in the group. The caller does not have to hang up and place another Intercom call if the first extension called is unavailable. Department Step Calling also allows an extension user to cycle through the members of a Department Group.

Dial Number Preview

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Dialing Number Preview lets a display keyset user dial and review a number before the system dials it out. Dialing Number Preview helps the user avoid dialing errors.

Dial Pad Confirmation Tone

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

For an extension with Dial Pad Confirmation Tone enabled, the user hears a beep each time they press a key. This is helpful for Intercom calls and Dial Pulse trunk calls, since these calls provide no Call Progress tones.

Dial Tone Detection

Aspire S	Aspire M/L/XL
• Available	• Available.
• Tone Detection Setup is available.	• Tone Detection Setup is available.
• The Next Trunk in Rotary if No Dial Tone option is available.	• The Next Trunk in Rotary if No Dial Tone option is available.

Description

If a trunk has Dial Tone Detection enabled, the system monitors for dial tone from the telco or PBX when a user places a call on that trunk. If the user accesses the trunk directly (by pressing a line key or dialing #9 and the trunk's number), the system will drop the trunk if dial tone does not occur. If the user access the trunk via a Trunk Group (by dialing a trunk group code or automatically through a feature like Last Number Redial), the system can drop the trunk or optionally skip to the next trunk in the group. Refer to the chart under Programming below for more.

Dial Tone Detection is available for the following features:

- Automatic Route Selection
- Abbreviated Dialing
- Central Office Calls, Placing
- Last Number Redial
- Loop Keys (outbound)
- Save Number Dialed
- T1 Trunking (with ANI/DNIS Compatibility)
- Tie Lines
- Trunk Group Routing
- Trunk Groups

Direct Inward Dialing (DID)

Aspire S	Aspire M/L/XL
• Available with software 2.21+ - 20 DID Translation Tables can be divided between 2000 entries.	• Available - 20 DID Translation Tables can be divided between 2000 entries.
• DID routing can follow ring group programming on transfers for busy/no answer calls.	• DID routing can follow ring group programming on transfers for busy/no answer calls with software 1.11+.
• Call by Time Schedule feature requires 4.93+.	• Call by Time Schedule feature requires 4.93+.
• Timer Class of Service requires software 5.91+	• Timer Class of Service requires software 5.91+

Description

Direct Inward Dialing (DID) lets outside callers directly dial system extensions. DID saves time for callers who know the extension number they wish to reach. To place a DID call, the outside caller dials the local exchange (NNX) and additional digits to ring the telephone system extension. For example, DID number 926-5400 can directly dial extension 400. The caller does not have to rely on attendant or secretary call screening to complete the call.

Note: Direct Inward Dialing requires DID service from telco.

In addition to direct dialing of system extensions, DID provides:

- DID Dialed Number Translation
- Flexible DID Service Compatibility
- DID Intercept
- DID Camp-On

DID Dialed Number Translation

DID allows different tables for DID number translation. This gives you more flexibility when buying DID service from telco. If you can't buy the exact block of numbers you need (e.g., 301-556), use the translation tables to convert the digits received. For example, a translation table could convert digits 501-756 to extension numbers 301-556.

The Aspire system has 2000 DID Translation Table entries that you can allocate among the 20 DID Translation Tables. There is one translation made in each entry. For a simple installation, you can put all 2000 entries in the same table. For more flexibility, you can optionally distribute the 2000 entries among the 20 tables.

In addition to number conversion, each DID Translation Table entry can have a name assigned to it. When the DID call rings the destination extension, the programmed name displays.

Flexible DID Service Compatibility

You can program the system to be compatible with three and four digit DID service. With four digit service, the telco sends four digits to the system for translation. With three digit service, the telco sends three digits to the system for translation. Be sure to program your system for compatibility with the provided telco service. For example, if the telco sends four digits, make sure you set up the translation tables to accept the four digits.

The system is compatible with Dial Pulse (DP) and DTMF DID signaling. DID trunks can be either wink start or immediate start.

DID Camp-On

DID Camp-On sets what happens to DID calls to busy extensions when you have Busy Intercept disabled. With DID Camp-On enabled, a call to a busy extension camps-on for the DID Ring No Answer Time interval. It then diverts to the programmed DID Intercept extension ring group or Voice Mail. Without DID Camp-On, the caller to the busy extension just hears busy tone.

DID Routing Through the VRS

DID calls can optionally route through the VRS. The DID caller hears an initial Automated Attendant Greeting explaining their dialing options. If the caller mis-dials, they can hear a second greeting with additional instructions. For example, the first Automated Attendant Greeting can be, “Thank you for calling. Please dial the extension number you wish to reach or dial 0 for the operator.” If the caller inadvertently dials an extension that doesn’t exist, they could hear, “The extension you dialed is unavailable. Please dial 0 for assistance or dial # to leave a message so we can call you back.”

You assign Automated Attendant greetings (i.e., VRS Messages) to the numbers in each Translation Table. This provides you with extensive flexibility when determining which greetings the system should play for which dialed numbers. You could, for example, set up 926 5401 through 926 5449 to route to extensions 301- 349, and have 926 5450 route to the automated attendant.

The system allows an extension to be defined as a 1-digit number which can be dialed by the outside caller on a DID/DISA trunk using the VRS. The outside caller is able to access to desired extension/department group by dialing only 1 digit after the system answers the call. If the same number is used as the first digit of an extension number as well as the 1-digit access code for DID/DISA, the outside caller will not be able to access the extension.

Example: If ‘2’ is defined as a 1-digit access code to department group 300, outside callers cannot access extensions 200-299 directly.

SMDR Includes Dialed Number

The SMDR report can optionally print the trunk’s name (entered in system programming) or the number the incoming caller dialed (i.e., the dialed DID digits). This gives you the option of analyzing the SMDR report based on the number your callers dial. (This option also applies to an ISDN trunk as well.)

DID Intercept

DID Intercept automatically reroutes DID calls under certain conditions. There are three types of DID Intercept:

- **Vacant Number Intercept**
  If a caller dials an extension that does not exist or mis-dials, Vacant Number Intercept can reroute the call to the programmed DID Intercept extension ring group or Voice Mail. Without Vacant Number Intercept, the caller hears error tone after mis-dialing.

- **Busy Intercept**
  Busy Intercept determines DID routing when a DID caller dials a busy extension. If Busy Intercept is enabled, the call immediately routes to the programmed DID Intercept extension ring group or Voice Mail. If Busy Intercept is disabled, the call follows DID Camp-On programming (see below).

- **Ring-No-Answer Intercept**
  Ring-No-Answer Intercept sets the routing options for DID calls that ring unanswered at the destination extension. With Ring-No-Answer Intercept enabled, the unanswered call reroutes to the DID Intercept extension ring group or Voice Mail after the DID Ring-No-Answer Time interval. If Ring-No-Answer Intercept is disabled, the unanswered call rings the destination until the outside caller hangs up.

Delayed DID

Delayed DID allows a user a pre-programmed amount of time to answer a call. If the call is not answered within this time period, the system automatically answers the call. An outside party will hear a voice message, music, or dial tone according to the following conditions:

- If a VRS is installed, the system sends a pre-recorded message from the VRS.
- If a customer-provided audio system (example: tape recorder) is connected, an error message or music can be played for the caller.
- If there is no equipment connected for an announcement, the system sends a unique dial tone to the outside caller.

This feature is not available for the normal incoming call on ISDN trunks.

DID Intercept Destination for Each DID Number

With this feature the system allows you to program a DID Intercept destination for a DID number which receives no answer or busy call. The system can be programmed to use a trunk ring group, the VRS or the voice mail as the programmed destination. Each vacant number intercept for a DID number can have two destinations. The first destination is for an invalid DID number, busy or no answer extension. The second destination is for a no answer trunk ring group.

For busy or no answer intercept calls, a third destination can be defined in Program 22-12. If the first and third destinations are programmed but the second destination is not, the incoming call goes to the third destination after the first destination. If the first and second destinations are not programmed, but the third destination is, the call goes directly to the third destination.

This feature works for DID trunks with a trunk service type 1 in Program 22-02. Other types of trunks may use the DID table, but the DID intercept feature is not yet supported.

With the DID Intercept for each DID number feature, when the primary destination (Program 22-11-05) is set to Voice Mail, the Voice Mail protocol is:

1. Busy Intercept = Forward Busy
2. Ring-No-Answer Intercept = Forward RNA

When the secondary destination (Program 22-11-06) is set to Voice Mail, the Voice Mail protocol is based on the first destination's routing. When the incoming call is forwarded to the first destination by a busy intercept, the Voice Mail protocol will be that it forwards busy calls. When the incoming call is routed to the first destination by a ring-no-answer intercept, the protocol will be that it forwards ring-no-answer. The Voice Mail will transfer the calls to the mailbox number defined in Program 22-11-02.

Note: Any valid DID number must be entered in the DID table (Program 22-11). If a valid DID number is not entered, there will be no ring destination for any incoming calls to that number (the calls will not ring any extension in the system).

Calls Can Follow Ring Group Programming for Transferring Calls

An option has been added to Program 22-11 which allows you to determine if the DID routing should use the programmed ring group entry in Program 22-12-01 when transferring calls from a busy or no answer number.

Call by Time Schedule Feature Added

For every DID number programmed, the DID Call by Time Schedule feature allows each of the 8 different time patterns (defined in Program 22-17) to be assigned a different destination, following the DID conversation table in Program 22-11. This time pattern can also be selected manually, using a service code. The time pattern used for this option is separate from the 8 day/night time modes defined in Program 12-02.

Notes:

- When the time pattern is changed manually, the following conditions apply:
  - Manually selecting the time pattern temporarily overrides the entries in Program 22-17. However, when a time pattern changes with the time schedules in Program 22-17, the pattern applied by the manual change is cancelled.
  - If the time pattern is manually selected and the system is reset, once the system restarts, the manually selected time pattern is still in affect.
- When a time pattern is set from 00:00 to 00:00 and Program 22-11 is defined, the system will always follow the setting in Program 22-11. To prevent the system from using Program 22-11, it must be undefined.
- If the destination defined in Program 22-17 is not found, the system then checks the setting in Program 22-11.
- If an incoming DID call is unanswered/busy, it will follow the entries in Program 22-12 and 22-13 for the trunk group based on the time the no answer/busy indication was received.

3

Federal Communications Commission DID Requirements

Allowing this equipment to operate in a manner that does not provide proper answer supervision signaling is in violation of Part 68 rules.

This equipment returns answer supervision to the Public Switched Telephone Network when the DID trunk is:

- Answered by the called station
- Answered by the attendant
- Routed to a recorded announcement that can be administered by the CPE user
- Routed to a dial prompt

This equipment returns answer supervision on all DID calls forwarded back to the Public Switched Telephone Network. Permissible exceptions are when:

- A call is unanswered
- A busy tone is received
- A reorder tone is received

Direct Inward Line (DIL)

Aspire S	Aspire M/L/XL
• Available - 8 trunks and 8 Department Groups. With software prior to 2.50 - 26 extensions and 24 virtual extensions. With 2.50+, 50 extensions and 24 virtual extensions.	• Available. **Aspire M/L:** 200 trunks, 64 Department Groups, 256 extensions, and 256 virtual extensions. **Aspire XL:** 200 trunks, 64 Department Groups, 384 extensions, and 256 virtual extensions.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.

Description

A Direct Inward Line (DIL) is a trunk that rings an extension, virtual extension or Department Group directly. Since DILs only ring one extension or group (i.e., the DIL destination), employees always know which calls are for them. For example, a company operator can have a Direct Inward Line for International Sales Information. When outside callers dial the DIL's phone number, the call rings the operator on the International Sales line key. The DIL does not ring other extensions.

DIL Delayed Ringing

Extensions in a Ring Group can have delayed ringing for another extension's DIL. If the DIL is not answered at its original destination, it rings the DIL No Answer Ring Group. This could help a Technical Service department, for example, that covers calls for an Inside Sales department. If the Inside Sales calls are not answered, they ring into the Technical Service department.

Direct Inward System Access (DISA)

Aspire S	Aspire M/L/XL
• Available - 15 users, 15 DISA Classes of Service and 8 trunks.	• Available - 15 users, 15 DISA Classes of Service and 200 trunks.
• Trunk Disconnect Continue and Disconnect Codes require software 4.0E+.	• Trunk Disconnect Continue and Disconnect Codes require software 4.0E+.
• Remote Setup with DISA requires software 4.93+.	• Remote Setup with DISA requires software 4.93+.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.

Description

DISA permits outside callers to directly dial system extensions, trunks and selected features. This could help an employee away from the office that wants to directly dial co-workers or use the company's trunks for long distance calls. To use DISA, the employee:

- Dials the telephone number that rings the DISA trunk
- Waits for the DISA trunk to automatically answer with a unique dial tone

- Dials the 6-digit DISA password (access code)
- Waits for a second unique dial tone
- Accesses a system trunk, uses a selected feature or dials a system extension

DISA calls ring system extensions like other outside calls. If an extension has a line key for the DISA trunk, the call rings that key. If the extension does not have a line key, the call rings an idle CALL key.

You can set DISA operation differently for each Night Service mode. For example, a trunk can be a normal trunk during the day and a DISA trunk at night. You can also set the routing for DISA trunks when the caller dials a busy or unanswered extension, dials incorrectly or forgets to dial.

DISA Class of Service

DISA Class of Service provides features and dialing restrictions for DISA callers. This allows you to control the capabilities of the DISA callers dialing into your system. When a DISA caller first accesses the system, they must enter a DISA password before proceeding. The system associates the password entered with a specific user number, which in turn has a Class of Service. If the Class of Service allows the action (such as making outgoing trunk calls), the call goes through. If the DISA Class of Service doesn't allow the action, the system prevents the call. The DISA Class of Service options are:

- **Trunk Group Routing/ARS Access**
  When a DISA caller dials into the system, they may be able to dial 9 and place outside calls. Any toll charges are incurred by the system. The call follows the system's Trunk Group Access or Automatic Route Selection - whichever is enabled.
- **Trunk Group Access**
  DISA callers may be able to access a specific trunk group for outgoing calls through the system. To access a Trunk Group, the user dials Service Code 804 followed by the Trunk Group number (e.g., 1). This allows the DISA caller to place an outgoing call over the selected group. Trunk Group Access bypasses the system's Trunk Group Routing/ARS/Trunk Access Maps. As with dial 9 access, any toll charges are incurred by the system. Also see Direct Trunk Access below.
- **Common Abbreviated Dialing**
  The system's Common Abbreviated Dialing bins may be available to DISA callers. This could save the DISA caller time when dialing.
- **Operator Calling**
  A DISA caller may be able to dial 0 for the system's operator.
- **Paging**
  Internal and External Paging may be available to DISA callers. This allows co-workers in adjacent facilities, for example, to broadcast announcements to each other.
- **Direct Trunk Access**
  DISA callers may be able to select a specific trunk for outgoing calls through the system. To directly access a trunk, the user dials Service Code #9 followed by the trunk's number (e.g., 001). This allows the DISA caller to place an outgoing call over the selected trunk. Direct Trunk Access bypasses the system's Trunk Group Routing/ARS/Trunk Access Maps. As with dial 9 access, any toll charges are incurred by the system. Also see Trunk Group Access above.
- **Call Forward**
  DISA callers may be able to set Call Forwarding to redirect an extension's calls to another extension. Call Forwarding ensures that the user's calls are covered when they are away from their work area.
- **DISA/Tie Trunk Barge In**
  The DISA/Tie Trunk Barge In option allows a tie line caller to break into another extension's established call. This sets up a three-way conversation between the intruding party and the two parties on the initial call.

DISA Toll Restriction
The digits a DISA caller dials for an outgoing call may be subject to the system's Toll Restriction. For example, Toll Restriction can prevent users from dialing a 1-900 service. When an incoming DISA caller tries to use system trunks to dial 1-900, Toll Restriction will deny the call.

DISA Operating Modes
The DISA Operating Modes determine what happens when a DISA caller forgets to dial, calls a busy or unanswered extension or dials incorrectly. The system can either drop the call or send it to a preset Ring Group (called a the DISA Transfer Destination).

Department Calling with Overflow Message
If a DISA caller dials a busy Department Calling Group, the system can periodically play the voice prompt, *"Please hold on. All lines are busy. Your call will be answered when a line becomes free."* while the caller waits. The interval between the voice prompts is the DISA Overflow Message Time. When an extension in the Department Group becomes available, the call automatically goes through. If the Department Calling Group remains busy past the DISA No Answer Time, the DISA call routes to the overflow destination or disconnects. (What happens to the unanswered call is set by the DISA Operating Mode). The Overflow Message requires a VRS.

Warning Tone for Long DISA Calls
You can set up the system to provide a warning tone to DISA callers that have been on a call too long. The warning tone can be just a reminder (which the caller can ignore) or can be followed by a forced disconnect of the call. When the DISA caller hears the warning tone, they have the option of dialing a code to continue the conversation or disconnect.

Trunk Disconnect Continue/Disconnect Codes Added
The software enhances the forced trunk release option with the Tandem Trunking and DISA features. With older software, with **Program 24-02-07 : System Options for Transfer - Forced Release for Trunk-to-Trunk Transfer** or **Program 25-07-08 : System Timers for VRS/DISA - DISA Long Conversation Disconnect** enabled, users were not provided with an option to continue the call, if required.

With software 4.0E+, users can be provided with the option to use a Continue or Disconnect service code. The Continue service code will extend the conversation a programmed length of time. If the user enters the Disconnect service code, the call will be disconnected immediately.

Remote Feature Setup with DTMF
An option may be available which can be used to remotely set various Aspire functions for the specified extension by dialing the extension number and service code using a DISA line. This option is available for keysets, single line telephones, and IP telephones.

When the outside caller, using an analog or ISDN trunk, places a call to a DISA line and dials the service code for this function, if the VRS is installed, the system will respond with a fixed message prompting the entry of the extension number ("Please dial the extension number."). After the outside caller dials the desired extension number, the system will respond with another fixed message ("Please enter the required Service Code."), then the outside caller dials the required service code to set/cancel the function.

The DISA feature must be enabled for this function.

Notes:

- While the outside caller is setting the function via DISA, no one can use the extension which is being set.
- The outsider caller can not set/cancel a function via DISA when the selected extension is being used except during incoming ringing (including incoming ACD calls). If the extension is busy, the call will be terminated.
- The VRS is required to send the fixed messages heard during the feature setup, but the feature can be used without the prompts.

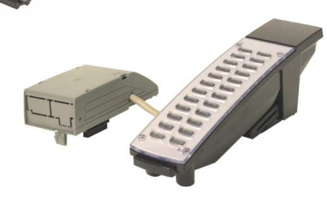
Direct Station Selection (DSS) Console

Aspire S	Aspire M/L/XL
• Available.	• Available.
• 4 110-Button DSS Consoles maximum (4 per extension).	• 32 110-Button DSS Consoles maximum (4 per extension).
• 24 24-Button DSS Consoles maximum (1 per extension).	• 256 24-Button DSS Consoles maximum (1 per extension).
• DSS Lamping programming is system wide.	• DSS Lamping programming is system wide with 1.02+.
• DSS Lamping affects DSS/Hotline keys for keysets with software 4.0E+.	• DSS Lamping affects DSS/Hotline keys for keysets with software 4.0E+.
• ACD is not available on Aspire S so the ACD agent/non-ACD agent lamping is not applicable.	• ACD agent and non-ACD agent lamping on a DSS Console requires software 4.0E+.

3

Description

The DSS Consoles (110-Button DSS: P/N 0890051 or 0890052; 24-Button DLS: P/N 0890053 or 0890054) gives a keyset user a Busy Lamp Field (BLF) and one-button access to extensions, trunks and system features. This saves time for users that do a lot of call processing (e.g., operators or dispatchers). The DSS Console simplifies:



- Calling extensions and Door Boxes
- Placing, answering and transferring outside calls
- Making an External or Internal Page
- Switching the Night Service mode
- Activating DSS Console Alternate Answer

The DSS Console also provides DSS Console Alternate Answer. This lets a keyset user with a DSS Console quickly reroute their calls to a co-worker. When the user places their console off-duty (by pressing the ALT. key), their calls route automatically to the programmed co-worker. Transferred and dial “0” calls ring both DSS Consoles and, if the VRS is installed, the main operator hears the message, “Your calls have been forwarded”. Central office calls ring both consoles and no message is heard by the operator.

You can also program the DSS Console keys to store Service Codes codes (up to 29 digits long). This provides the DSS Console user with many of the features available on One-Touch and Programmable Feature Keys. The DSS Console keys can optionally store additional associated digits after the Service Code. For example, storing 8141 under a DSS Console key accesses Trunk Group 1 when the console user presses the key.

The maximum number of 110-Button DSS Consoles allowed per system is 4 (Aspire S) or 32 (Aspire M/L/XL). Up to 24 (Aspire S) or 256 (Aspire M/L/XL) 24-Button DSS Consoles are possible. Each extension can have up to 4 110-Button DSS Consoles and/or only one 24-button DLS console. An extension can, however, have both a 24-button DLS and 110-button DSS console(s) installed. The 110-Button DSS Console requires an available extension port - the 24-Button DLS Console does not.

24-Button DLS (Direct Line Select): DSS/DLS Service Code

Using Programming Function Keys, you can determine whether the keys on a 24-button DLS console (P/N 0890053 or 0890054) are used as DSS (direct station selection) or DLS (direct line selection) keys.

To prevent lamping problems when reassigning DLS Console keys, it is recommended that you clear an extension's programmed key before reassigning it (Program 30-03: Enter key to be cleared + FLASH key [If using PCPro/WebPro, delete the entry and upload the change to the system before proceeding]). Without clearing an extension's key first, your DSS Console may not show the correct lamping, although the DSS function will work correctly.

If you are programming the system from the extension to which the DLS Console is connected, either by phone or using the PC Program, you may need to unplug the DLS and plug it back in to reset the console's lamping.

DSS Lamping Changed to Apply to All Consoles

Software has been changed to apply settings in **Program 30-05 : DSS Console Lamp Table** to all system consoles. Previously, the settings in this program defined for each DSS Console.

DSS Lamping Table Changed to Apply to DSS/Hotline Keys for Keysets

Software has been changed to apply settings in **Program 30-05 : DSS Console Lamp Table** to DSS and Hotline keys on keysets as well as DSS Consoles.

ACD/Non-ACD Agent DSS Lamping Available

With the Aspire M/L/XL system and software 4.0E+, a new option is available in Program 30-05-01 which allows a non-ACD DSS console to lamp indicating the status of both non-ACD agents as well as ACD agents. Previously, the console would only display one type or the other.

Directed Call Pickup

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Directed Call Pickup permits an extension user to intercept a call ringing another extension. This allows a user to conveniently answer a co-worker's call from their own telephone. With Directed Call Pickup, an extension user can pick up:

- Trunk calls (i.e., Ring Group calls)
- Direct Inward Lines
- Transferred trunk calls
- Transferred Intercom calls
- Ringing and voice-announced Intercom calls

Directory Dialing

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Directory Dialing allows a display or Super Display keyset user to select a co-worker or outside call from a list of names, rather than dialing the phone number. There are four types of Directory Dialing:

- **ABBc** - Company (Common) Abbreviated Dialing
- **ABBg** - Department (Group) Abbreviated Dialing
- **EXT.** - Co-worker's extensions
- **OneT** - Personal Abbreviated Dialing (One-Touch Keys)

Display Messaging, Selectable

Refer to the **Selectable Display Messaging** (page 184) for information on this feature.

Distinctive Ringing, Tones and Flash Patterns

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Distinctive Ringing, Tones and Flash Patterns provide extension users with audible and visual call status signals. This lets users tell the types of calls by listening to the ringing/tones and watching the keys. It also helps users monitor the progress of their calls. In addition, Distinctive Ringing lets keyset users customize their Intercom and trunk call ringing. This is helpful for users that work together closely. For example, if several co- workers set their keysets to ring at different pitches, the co-workers can always tell which calls are for them.

You can also customize the tones the system uses for splash tone, confirmation tone, trunk ring tone, Intercom ring tone and Alarm ring tone.

Do Not Disturb

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Class of Service option for allowing DND requires software 2.63+.	• Class of Service option for allowing DND requires software 2.63+.
• DND activation while on a call is possible with 2.63+ software.	• DND activation while on a call is possible with 2.63+ software.

Description

Do Not Disturb blocks incoming calls and Paging announcements. DND permits an extension user to work by the phone undisturbed by incoming calls and announcements. The user can activate DND while their phone is idle or while on a call. Once activated, incoming trunk calls still flash the line keys. The user may use the phone in the normal manner for placing and processing calls.

There are five Do Not Disturb options available at each extension:

1 = Incoming trunk calls blocked
2 = Paging, incoming Intercom, Call Forwards and transferred trunk calls blocked
3 = All calls blocked
4 = Incoming Call Forwards blocked
0 = Do Not Disturbed canceled

Keyset users be able to activate or deactivate Do Not Disturb while on a call, depending on their software version. With older software, the keyset user could only enable DND when the phone was idle or when dial tone was heard. This option is not available for single line telephones.

Door Box

Aspire S	Aspire M/L/XL
• Available - 4 Door Boxes and 6 Chime Tones.	• Available - 8 Door Boxes and 6 Chime Tones.

Description

The Door Box is a self-contained Intercom unit typically used to monitor an entrance door. A visitor at the door can press the Door Box call button (like a door bell). The Door Box then sends chime tones to all extensions programmed to receive chimes. To answer the chime, the called extension user just lifts the handset. This lets the extension user talk to the visitor at the Door Box. The Door Box is convenient to have at a delivery entrance, for example. It is not necessary to have company personnel monitor the delivery entrance; they just answer the Door Box chimes instead. Any number of system extensions can receive Door Box chime tones.

Each Door Box has a pair of normally open relay contacts that can connect to an electric door strike. Use these contacts to remotely control the entrance door. After answering the Door Box chimes, a keyset user can press FLASH to activate the Door Box contacts. This in turn releases the electric strike on the entrance door. The system can have up to eight Door Boxes.

Dual Line Appearance

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Each keyset has two line appearance keys (CALL1 and CALL2) for placing and answering calls. These line appearance keys, assigned to the extension's number, simplify operations for busy users. For example, the user can easily process a new call on one appearance with a call in progress on the other.

Dual OPX/2-OPX

Refer to the **Single Line Telephones, Analog 500/2500 Sets** (page 186) for information on this feature.

E911 Compatibility

Aspire S	Aspire M/L/XL
• Available.	• Available.
• IP phones following ARS Class of Service Matching with networked systems to call local authorities with 911 call requires software 2.63+.	• IP phones following ARS Class of Service Matching with networked systems to call local authorities with 911 call requires software 2.63+.

Description

E911 Compatibility ensures that emergency calls always get through. If an emergency occurs, a user simply goes to any phone, lifts the handset and dials 911. The system's built-in E911 compatibility places the emergency call even if the user forgets to dial an access code or press a line key. The E911 capabilities include:

- **Attendant Notification**
  The attendant receives a notification each time a co-worker dials an emergency 911 call. This notification is the co-worker's name and number display optionally accompanied by an audible alarm. Notification occurs regardless of whether the attendant is idle or busy on a call. You can optionally extend this capability to other supervisory extensions as well.

- **Emergency Routing**
  When an extension user dials 911, the system can automatically find a trunk for the call. The system can choose a route to which the user normally does not have access. If all normal routes are busy, the system can even disconnect an active call and place the emergency call. E911 Compatibility uses the flexibility of the Automatic Route Selection Call Route Options to route 911 emergency calls (even in systems in which ARS is not enabled).
  - **E911 Outgoing Dialing**
    The E911 calls follow the trunk group route programming. It is possible to use the flexibility of the Automatic Route Selection Call Route Options for additional routing options.
  - **Forced Disconnect Follows Timer to Disconnect Call**
    When all lines in the programmed route are busy and the system must drop a call in order to place a 911 call, the system will wait the amount of time set in Program 81-01 before disconnecting the call.

- **Compatibility with Customer Provided E911 Equipment**
  The system can automatically send a 911 call to customer-provided E911 equipment (such as the Proctor 911 ANI-LINK System II). The E911 equipment will dial emergency service and provide the caller's extension number to the emergency personnel. When using this type of equipment, the system must be programmed to send E911 calls to the trunk connected to the E911 box. This is done by assigning the trunk to the E911 trunk group and using ARS to route all 911 calls to that port. The Dial Treatment must then be set to send the extension number and '911' to that port. This is usually accomplished using XRE in the Dial Treatment, but check the requirements of your E911 box.

- **Calling Party Identification**
  With ISDN installed, the system can provide the calling party's telephone number and extension number. No additional customer-provided 911 equipment is required.

Uninstalled Trunks in Trunk Group Prevent Call from Dialing Out
By system default, all trunks in **Program 14-05-01 : Trunk Group** are in group 1. When placing a 911 call, the system will try to access the trunks defined in the group. If the trunks do not exist, the call will not dial out. In order for E911 to function correctly, remove any uninstalled trunks from the trunk group.

The system software will be changed in a future release to ignore trunks which have no voltage present on the PCB.

Using Dial Number Preview with 9+911 Call Will Not Dial Out When All Trunks Busy
If **Program 21-01-12 : System Options for Outgoing Calls, Dial 911 Routing Without Trunk Access** is set to "0" (trunk access code required), when using the Dial Number Preview feature and dialing 9+911, if all trunks are busy, the user will hear a busy signal and the call will not dial out.

If option 21-01-12 is set to "1" (trunk access code not required) and using Dial Number Preview, 911 is dialed, the system will disconnect a trunk and dial the call.

Dial Number Preview is when a telephone number is first dialed (previewing the number in the display) then pressing the CALL or line key to place the call.

Networked IP Phones Can Follow ARS Class of Service Matching to Call Local Authorities
When using IP telephones at a remote site (Site B) which are registered to the main system (Site A), you can use the ARS Class of Service Matching feature to route 911 calls to the local authorities at a remote location. Without this programming, since the phones are registered to the main site, when 911 calls are placed by IP phones, the local authorities at the main site (Site A) would be called.

External Alarm Sensors

Aspire S	Aspire M/L/XL
• **This is a future item and is not currently available.**	• **This is a future item and is not currently available.**

This is a future item and is not currently available.

Flash

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Flash allows an extension user to access certain CO and PBX features by interrupting trunk loop current. Flash lets an extension user take full advantage of whatever features the connected telco or PBX offers. You must set the Flash parameters for compatibility with the connected telco or PBX.

Flexible System Numbering

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Flexible System Numbering lets you reassign the system's port-to-extension assignments. This allows an employee to retain their extension number if they move to a different office. In addition, factory technicians can make comprehensive changes to your system's number plan. You can have factory technicians:

- Set the number of digits in internal (Intercom) functions. For example, extension numbers can be up to eight digits long.
- Change your system's Service Code numbers
- Assign single digit access to selected Service Codes

Talk to your sales representative to find out if this program is available to you.

You can also use Flexible System Numbering to change the system's Trunk Group Routing code. Although the default code of 9 is suitable for most applications, you can alter the code if needed.

Forced Trunk Disconnect

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Forced Trunk Disconnect allows an extension user to disconnect (release) another extension's active outside call. The user can then place a call on the released trunk. Forced Trunk Disconnect lets a user access a busy trunk in an emergency, when no other trunks are available. Maintenance technicians can also use Forced Trunk Disconnect to release a trunk on which there is no conversation. This can happen if a trunk does not properly disconnect when the outside party hangs up.

CAUTION
Forced Trunk Disconnect abruptly terminates the active call on the line. Only use this feature in an emergency and when no other lines are available.

Group Call Pickup

Aspire S	Aspire M/L/XL
• Available - 8 Call Pickup Groups.	• Available - 64 Call Pickup Groups.

Description

Group Call Pickup allows an extension user to answer a call ringing an extension in a Pickup Group. This permits co-workers in the same work area to easily answer each other's calls. The user can intercept the ringing call by dialing a code or pressing a programmed Group Call Pickup key. If several extensions within the group are ringing at the same time, Group Call Pickup intercepts the call based on the extension's priority within the Pickup Group.

With Group Call Pickup, a user can intercept the following types of calls:

- A call ringing the user's own pickup group
- A call ringing another pickup group when the user knows the group number
- A call ringing another pickup group when the user doesn't know the group number

Group Listen

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Group Listen permits a keyset user to talk on the handset and have their caller's voice broadcast over the telephone speaker. This lets the keyset user's co-workers listen to the conversation. Group Listen turns off the keyset's Handsfree microphone so the caller does not pick the coworker's voices during a Group Listen.

Handsfree and Monitor

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

(instead of the handset). Handsfree is a convenience for workers who don't have a free hand to pick up the handset. For example, a terminal operator could continue to enter data with both hands while talking on the phone.

The system provides three variations of Handsfree operation:

Handsfree	User can place and answer calls by pressing SPK instead of using the handset.
Automatic Handsfree	User can press a line or line appearance key without first lifting the handset or pressing SPK. An extension can have Automatic Handsfree for just outgoing calls or both outgoing calls and incoming line/loop key calls. Automatic Handsfree can also be used with the Call Coverage or Park features. Normally, extensions without speakerphones should have Automatic Handsfree for outgoing calls only.
Monitor	User can place a call without lifting the handset, but must lift the handset to speak.

Handsfree Answerback/Forced Intercom Ringing

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Handsfree Answerback permits an extension user to respond to a voice-announced Intercom call by speaking toward the phone, without lifting the handset. Like Handsfree, this is a convenience for workers who don't have a free hand to pick up the handset.

Headset Operation

Aspire S	Aspire M/L/XL
• Available.	• Available.
• SLT Headset Operation available.	• SLT Headset Operation available with 1.02+.

Description

A keyset user can utilize a customer-provided headset in place of the handset. Like using Handsfree, using the headset frees up the user's hands for other work. However, Headset Operation provides privacy not available from Handsfree.

An extension with a headset has two options for when it appears busy to incoming callers. The headset extension can be:

- Busy to incoming callers when only one extension appearance is busy (i.e., Off-Hook Signaling prevented)
  OR
- Busy to incoming callers only when both extension appearances are busy (i.e., Off Hook Signaling allowed)

As the headset plugs into a separate jack on the bottom of the phone, the handset can still be connected to the phone. This provides you with the option to use the handset, headset or the speakerphone for calls.

Examples of compatible headsets are the:

- Polaris Supra Monaural Noise Cancelling, P/N 750036
- Polaris Supra Binaural Noise Cancelling, P/N 750033
- Polaris Encore Binaural Noise Cancelling, P/N 750035

Headset Operation for SLT Headset Operation

The ability for single line telephone users to use the Headset feature has been added. When a single line telephone with a headset receives an incoming call, the system can let the SLT user know of the incoming call by a notification tone in their headset.

Hold

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.

Description

Hold lets an extension user put a call in a temporary waiting state. The caller on Hold hears silence or Music on Hold, not conversation in the extension user's work area. While the call waits on Hold, the extension user may process calls or use a system feature. Calls left on Hold too long recall the extension that placed them on Hold. There are four types of Hold:

- **System Hold**
  An outside call a user places on Hold flashes the line key (if programmed) at all other keysets. Any keyset user with the flashing line key can pick up the call.
- **Exclusive Hold**
  When a user places a call on Exclusive Hold, only that user can pick up the call from Hold. The trunk appears busy to all other keysets that have a key for the trunk. Exclusive hold is important if a user doesn't want a co-worker picking up their call on Hold.
- **Group Hold**
  If a user places a call on Group Hold, another user in the Department Group can dial a code to pick up the call. This lets members of a department easily pick up each other's calls.
- **Intercom Hold**
  A user can place an Intercom call on Hold. The Intercom call on Hold does not indicate at any other extension.

With Automatic Hold enabled (Program 15-02-07), when the user is on an ***outside call using the handset***, the user can press a flashing line/loop key to answer an incoming call without disconnecting their first call. The first caller is automatically placed on hold. This feature does not work using handsfree or when the user is on an ICM and presses a flashing line/loop key (the ICM call is disconnected).

Hold Recall to Operator

Hold Recall to Operator enhances how the system handles calls that have been left on hold too long. With Hold Recall to Operator:

- A trunk call recalls the extension that placed it on Hold after the Hold/Exclusive Hold Recall time.
- The recalling trunk will ring the extension that placed it on Hold for the Hold/Exclusive Hold Recall Callback Time.
- After the Hold/Exclusive Hold Recall Callback Time, the trunk call will ring the operator.

Hold Recall to Operator applies to trunk calls placed on System Hold, Exclusive Hold and Group Hold. It does not apply to Intercom calls.

Hotel/Motel

Aspire S	Aspire
• Available with software 5.00+.	• Available with software 5.00+.

Description

The system can provide comprehensive hotel/motel services in addition to the features normally available to business users. Hotel/Motel features include:

- **Wake Up Call**
  A guest can set or cancel a wake-up call request. A hotel/motel employee with a keyset can also set or cancel a wake-up call for a room telephone.

- **Single Digit Dialing**
  Single Digit Dialing gives guests one-touch access to important Hotel/Motel services. They can just lift the handset and press a single key for:
  - Extensions such as the front desk, reservation services, housekeeping or the maitre d' of the restaurant.
  - Feature Access Codes for one button access to selected features and outside lines.
  - Voice Mail, so guests can leave requests even when service providers are unavailable.
  - A Department Calling Group allowing, for example, guests to reach the first available agent in the reservation desk group.

- **Message Waiting**
  A hotel/motel employee with a keyset can send a Message Waiting to a room telephone. The message lamp on the room telephone flashes until the guest answers the Message Waiting. (The DSS Console can show all the rooms that have messages waiting.)

- **Room-to-Room Calling Restriction**
  Prevent guests in one room from calling guests in another — a hotel/motel employee with a keyset can enable and disable room- to-room calling.

- **Toll Restriction (When Checked In)**
  An employee can change the Toll Restriction for a guest’s telephone. For example, the receptionist can enable long distance calling for each room telephone as the guests check in.

- **Room Status**
  To better manage room usage, an employee with a keyset can change the status of a room telephone, including:
  - Checked In
  - Checked Out
  - Maid Required
  - Maid in Room

- **Room Status with Printout**
  An employee's DSS Console can indicate the status of the hotel/motel rooms. Optionally, a printer connected to a CTA Module can print out room status reports to provide more detailed information:
  - Room Status (occupied, available, ready and to be cleaned)
  - Room Telephone Call and Toll Restriction Information
  - Do Not Disturb and Clean Up Extension List
  - Message Waiting Report
  - Wake-up Call No-Answer Report

- **DSS Console Monitoring**
  Your DSS Console provides unique one-touch room monitoring capabilities. Just press a button on your DSS Console to check a room's status. Or, see at a glance which rooms have Wake Up Calls set or messages waiting. In addition, you can still use your console for business mode features.

- **Do Not Disturb**
  A guest can enable and disable Do Not Disturb for their room telephone. In addition, a hotel/motel employee with a keyset can enable and disable Do Not Disturb for a specific room telephone. Do Not Disturb (DND) blocks the room telephone's incoming calls and Paging announcements.

- **Flexible Numbering Plan**
  To simplify dialing guests and services in your facility, customize your system to have room numbers match each phone's extension number. For example, if the rooms on the first floor are numbered 100-120, the corresponding room extensions can also be 100-120.

Hotline

Aspire S	Aspire M/L/XL
• Available - 512 Internal Hotline extensions.	• Available - 512 Internal Hotline extensions.
• DSS Lamping (Program 30-05-01) applied to keysets with 4.0E+ for DSS and Hotline keys.	• DSS Lamping (Program 30-05-01) applied to keysets with 4.0E+ for DSS and Hotline keys.

Description

Hotline gives a keyset user one-button calling and Transfer to another extension (the Hotline partner). Hotline helps co-workers that work closely together. The Hotline partners can call or Transfer calls to each other just by pressing a single key.

In addition, the Hotline key shows the status of the partner's extension

When the key is . . .	The extension is . . .
Off	Idle
On	Busy or ringing
Fast Flash	DND - All calls (option 3) or Intercom calls (option 2)
Double Wink On	ACD Agent logged onto the group
Wink Off	ACD Agent logged off

Hotline, External

Refer to the **Ringdown Extension, Internal/External** (page 181) for information on this feature.

i-Series Telephones

Aspire S	Aspire M/L/XL
• Not Available.	• Available with software 1.06+.

Description

Each 16DSTU PCB provides the Aspire system the ability to connect up to 16 i-Series telephones with each port supporting 1 B-channel. The system allows up to 15 DSTU PCBs (limited by load factor) to be installed, allowing up to 240 i-Series telephones to be connected. Each phone having a load factor of "3". The following i-Series phones are compatible with the Aspire system:

Model 2 922xx/926xx Series Keysets

32-Button Display Phone, P/N 92293B / 92293W / 92673
32-Button Standard Phone, P/N 92290B / 92290W / 92670
16-Button Display Phone, P/N 92373C / 92373W / 92573 / 92563
16-Button Standard Phone, P/N 92370B / 92370W / 92570 / 92560

Model 3 i-Series Keysets *

34-Button Display Phone, P/N 92783
28-Button Display Phone, P/N 92763
28-Button Standard Phone, P/N 92760
22-Button Display Phone, P/N 92753A / 92750A

** i-Series keysets do not have a gain setting database and will use the phone's initial setting*

In addition, the following hardware is also supported using DSTU ports:

- Remote Extender

The following i-Series hardware is **NOT** supported on the Aspire system:

- 2OPX
- 3ACI
- 900/900i/910i Cordless Phone
- DCI-A/B
- DCI-L
- Digital VANGARD Voice Mail
- DSLT
- DSS Consoles (24-Button or 110-Button)
- VAU
- Analog Module
- Data Module
- Off-Hook Voice Announce Module
- Speakerphone Module
- Super Display Phones

When using i-Series phones on the Aspire system, the following features are **NOT** supported:

- Directory Dial
- Super Display Operation
- Soft Keys
- Headset Key
- Changing Incoming CO and ICM Ring Tones (Program 11-11-20)
- Telephone System Programming (#*#*)
- Program 10-03 : PCB Setup (DSTU PCB has no programmable options)
- Program 90-17-01 : Display Firmware Version
- Program 90-07-01 : Extension Control

In addition to the above, the 92290x, 92670, 92370x, 92570, 92560, and 92760 phones do **NOT** support:

- Check Abandon Calls (CHECK + CALL2)
- Check Port/Name (CHECK + CALL1)
- Name Program (Service Code 800)
- Language Display (Program 15-02-01)
- Time and Display Modes (Program 20-02-07)

Call History/Caller ID Lists Do Not Provide Delete Function

With software 4.93-4.95, the display of the Call History/Caller ID list was enhanced to provide the ability to delete one or all numbers currently displaying in the Caller ID list.

With this change, however, when reviewing the numbers in the list, when the last number in the list was viewed, it would then scroll back to the first number - it did not stop. With this change, the i-Series phones did not have the option to clear the list by pressing CHECK + 9 (CHECK + 0 aborts the deletion).

This has been changed with this release so that the list will now stop at the last number when reviewing using an i-Series phone.

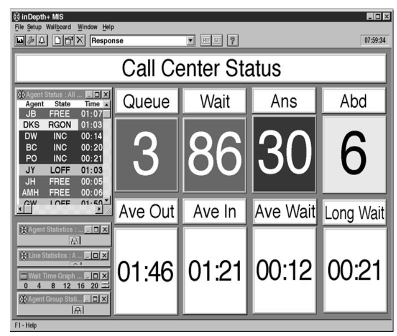
InDepth Lite, inDepth and inDepth+

Aspire S	Aspire M/L/XL
• Available, but not for ACD. Requires Aspire software 2.65+ and inDepth software 3.10.500.1 +.	• Available (Optional feature) - Requires inDepth's Aspire software 3.10.500.1 or higher and the Feature Upgrade PAL or Enhanced NTCPU when used with ACD. With the Basic NTCPU and no ACD, software 2.65+ is required.

Description

work in conjunction with the built in Aspire ACD. These ACD/MIS systems enhance the system ACD with real time statistics and reports on ACD group traffic patterns and usage. The inDepth and inDepth+ applications have an extensive set of user-configurable Real Time Windows and Reporter subsystems.

InDepth+ is the more comprehensive and capable of the ACD/MIS systems and offers:



- **Real Time Status Window**
  This window displays ACD agent status, state and connection state.
- **Real Time Statistics Window**
  The statistics window provides a visual performance summary for lines, agents and ACD Groups.
- **Call Queue and Wait Time Windows**
  These windows show the number of calls in queue, the longest wait time, as well as the number of calls answered and abandoned.
- **Wallboard Template**
  Use the wallboard template display to motivate and inform ACD agents through a dynamic display of real time statistics and messages.
- **Reporter**
  ACD administrators can create fully-configurable reports for display and printing.

Similar in many respects to inDepth+, inDepth is streamlined for more modest ACD applications. InDepth provides a single real time screen template, up to seven reports and can track report data for up to one full month. InDepth includes ACD/MIS features like Report View/Print and Audible/Visual Alarms, but excludes the Sub-Supervisor Positions and the Wallboard Support.

Hardware, Software and System Requirements

- Pentium 3 700 MHz or equivalent, 256 RAM
- Windows XP, Windows NT, Windows 2000 Professional, Windows 2000 Server, Windows 2003 Server
- SVGA mode (800 x 600)
- 2 GB Hard Drive Space
- Monitor
- CD-Rom Drive
- Network card - 10 Mbps
- Sound card
- 3 Serial Ports (depending on configuration)
- 1 Parallel Port

For more information, refer to the inDepth/inDepth+ Manual (P/N 0893230) for the specifics.

Intercom

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.

Description

Intercom gives extension users access to other extensions. This provides the system with complete internal calling capability.

Handsfree Answerback/Forced Intercom Ringing

Handsfree Answerback permits an extension user to respond to a voice-announced Intercom call by speaking toward the phone, without lifting the handset. Like Handsfree, this is a convenience for workers who don't have a free hand to pick up the handset. Refer to **Handsfree Answerback/ Forced Intercom Ringing** (page 134) for more.

Busy Status Display

When a display keyset user places an Intercom call to a busy extension, the details of the busy status (who it talking to the extension or which line is in use by the extension) can be displayed. The details of the trunk's busy status (the extension using the line) can be displayed after trying to access the trunk. This feature provides a user information which can determine whether they should use the Barge-In feature for the extension or trunk. This information automatically displays for a keyset once programmed.

Intercom Abandoned Call Display

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Intercom Abandoned Call Display shows a display keyset user a list of Intercom calls placed to them that they did not answer. This is a convenience if a user has to temporarily leave their desk. When they return, they can display the list to find out who called while they were out.

ISDN Compatibility

Aspire S	Aspire M/L/XL
• PRI Not Available.	• PRI Available.
• BRI S-Bus Available. BRI T-Bus Not Yet Available.	• BRI S-Bus Available. BRI T-Bus Not Yet Available.
	• This feature requires the Basic NTCPU (P/N 0891002) with the PAL upgrade or the Enhanced NTCPU (P/N 0891038) unless using software 4.0E or higher.
• CLIP display available with software 1.02+.	• CLIP display available with software 1.02+.
• Calling Party allowed for extension with software 1.04+.	• Calling Party allowed for extension with software 1.04+.
• Option for alert tone for S-Bus terminals calling busy extension available with 2.63+ software.	• Option for alert tone for S-Bus terminals calling busy extension available with 2.63+ software.
• T303 timer operation mode option available with 2.63+ software.	• T303 timer operation mode option available with 2.63+ software.
• Option for PB Back Tone Level available with 2.63+ software.	• Option for PB Back Tone Level available with 2.63+ software.
	• 2 B-Channel Transfer for PRI requires software 2.63+.
	• Alternate Carrier Access added to ISDN trunks with ARS requires software 2.64+. Refer to the **Automatic Route Selection** (page 144) for additional information.
• Ringback tone for BRI trunks requires software 4.04+.	• Ringback tone for BRI trunks requires software 4.04+.
• Calling Party Number option to define type requires software 4.08+.	• Calling Party Number option to define type requires software 4.08+.
• QSIG-Basic Call function requires software 5.96+.	• QSIG-Basic Call function requires software 5.96+.

Description

!! Important !!

Always check with your NEC's Technical Service Representative before setting up your ISDN application. Working together will ensure maximum compatibility and reliable ISDN performance.

SMDR Includes Dialed Number

The SMDR report can optionally print the trunk's name (entered in system programming) or the number the incoming caller dialed (i.e., the dialed ISDN digits). This gives you the option of analyzing the SMDR report based on the number your callers dial. (This option also applies to a DID trunk as well.)

Display Shows Why Caller ID is Not Available

With Caller ID enabled, the system will provide information for ISDN calls that do not contain the Caller ID information. If the Caller ID information is restricted, the telephone display will show "PRIVATE". If the system is not able to provide Caller ID information because the telco information is not available, then the display will show "OUT OF AREA".

Calling Party Number Notification

The system can provide calling party number notification for outgoing ISDN calls. When a call is made on an ISDN line by an extension, the system will send the identification for the extension placing the call, if it's programmed. If there is no Extension Calling Number assigned, the system will send the calling number for the ISDN trunk. If both the extension and trunk information is programmed, the extension information will be sent as it takes priority.

By telco regulation, some calling party information must be sent (either the main telephone number or a programmed name). Otherwise, the call can be sent as "Private", however, if the receiving system has anonymous call rejection enabled, the call will not go through.

When the option for calling party subaddress is on, the extension number will be sent as the subaddress information. Both the calling party number and calling party subaddress are sent in a SETUP message as the calling party information element and a calling party subaddress information element. Allow the system to send the subaddress by setting the following programs: 10-03-05=1, 15-01-04=1, 20-08-13=1, 21-13-01=enter number to be sent.

Calling Line Identification Presentation

A Class of Service option has been added which can be used to allow the Calling Party Number IE in the Setup Message.

Calling Party Allowed or Prevented for Extension

The system now allows the Calling Party Number for outgoing ISDN calls based on the extension's set up in **Program 15-01-04 : Basic Extension Data Setup - ISDN Caller ID**. If this option is to be enabled, then it must also be enabled for the BRI or PRI PCB in **10-03-05 : PCB Setup - CLIP Information**. A user can also press an ISDN Caller ID Block Programmable Function Key (code 63). When this key is pressed, all numbers dialed from the keyset will display on the far end as a "Private" call.

Alert Tone When S-Bus Terminal Calls Busy Extension

System software provides a program option for S-Bus terminals to determine what a user on an S-Bus terminal will hear when a busy extension is called. The system can be programmed so the user will hear an alert tone or be disconnected.

Release Message with Second T303 Timer
Depending on your software, the system can provide an program option for ISDN trunks can be used to determine whether or not a release message is sent when the second T303 timer expires.

Option Added for PB Back Tone Level
An option is available in system software which can be used to adjust the PB Back Tone level when calling an ISDN line.

Calling Party Number Option Added to Define Type
System software allows the number plan type to be defined in programming. Previously, this was not selectable and, depending on your software, was either set for "Unknown" or "National number in a ISDN numbering plan" to allow the calling party information to be passed to some telcos.

With software 4.08, this option was programmable using Programs 99-01-32 and 99-01-33, allowing the system to be adjusted for this setting. With software 4.93+, these programs have been replaced by 10-03-18 and 10-03-19.

QSIG-Basic Call (Non Transit PINX) Function Added
The standard for the ISDN interconnection of telecommunication systems is called QSIG (Signalling at the Q reference point). It is replacing or has already replaced signalling over E&M lines on leased lines. This type of ISDN-based signaling protocol can be used to connect the Aspire system to a 3rd party telephone system. The implemented functionality reaches from basic call control up to complete private network-wide call control and routing. Normally, between different telephone systems, only basic calls work.

With this software, the Aspire systems can now be attached to a QSIG network. The major difference between an BRI ISDN link and a QSIG link is the length of the Call Reference value. ISDN lines can now be marked in the system data by a new flag to indicate the length of the call reference value. To enable the QSIG mode for PRI or BRI trunks, **Program 10-03-22 : PCB Setup - QSIG Operation Mode** has been added. Set this option to "1" to enable QSIG or "0" to disable. By default, this option is disabled.

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Limitations
The feature QSIG is limited to an endpoint in a QSIG network. This applies especially to calls arriving via the QSIG network at the Aspire system.

The standard ECMA-143 names these call situations:

- Terminating PINX, if a call via QSIG is terminated at that PINX.
- Originating PINX, if a call originating at that PINX is routed via QSIG.
- Transit PINX, if a call is arriving via QSIG at that PINX at routed forward via QSIG (not supported).
- Incoming Gateway PINX, if a call incoming from public ISDN network is routed forward via QSIG.
- Outgoing Gateway PINX, if a call incoming via QSIG is routed outbound to an ISDN public network (not supported).

This feature is meant to implement the procedures needed for a Terminating PINX, an Origination PINX or an Incoming Gateway PINX. This software does not support the feature for a Transit PINX or an Outgoing Gateway PINX.

PINX stands for Private ISDN Network Exchange.

Incoming calls will route as if it had been received on a normal ISDN trunk. It will follow the data in **Program 22-09 : DID Basic Data Setup** for the number of digits, and **Program 22-11 : DID Translation Number Conversion** for a target destination. The software does not support dialing of additional digits to access a trunk at another PINX.

Functional Differences from Normal ISDN Operation

Below are described differences from normal ISDN operation, whether additional information elements are used or different or new procedures are applied.

- **CalledPartyNumber in SETUP Message**
  In any SETUP message, a CalledPartyNumber-IE which contains at least the first digit is mandatory. This must be assured by the system data, e.g. by means of F-Routing or ARS, which makes sure that at the moment the QSIG trunk is seized, at least one digit to be dialled at the trunk is available and is dialled to the trunk.
- **Party Category**
  Party category is an optional information element specifying whether the referred party is a normal extension, operator or an extension performing an emergency call.
- **Calling Number Plan, Called Number Plan**
  Within QSIG networks, a private number plan may be used and announced in the Called Party Number and Calling Party Number information elements by the Number Plan Indicator 9 (binary 1001). This can be set in Programs 10-03-18 through 10-03-19.

Relations to Standards

The feature makes uses of one or more of the following standards:

- ECMA 106 Signalling at the S-Reference Point
  This standard describes what happens if an ISDN terminal is attached to a PINX.
- ECMA 143 Inter Exchange Signalling Procedures and Protocol
  This standard describes the use and coding of the information elements clause 14.3 specifies the CR to be sent in two octets.
- ECMA 133 Reference configurations for PISN Exchanges (PINX)
  This standard describes configuration for PINX interconnections.

Primary Rate Interface (PRI)

The system is compatible with ISDN Primary Rate Interface (PRI) services. PRI services currently supported include:

- Basic PRI Call Control (BCC)
- Display of incoming caller's name and number when allowed by telco
- Routing in the system based on the number the caller dialed
- ISDN maintenance functions (such as In Service/Out of Service Messaging)
- Speech and 3.1 KHz audio

PRI capability requires the installation of T1/PRI Interface PCBs (0891009). Each PCB (also called a PRI circuit) provides 24 PRI channels (23B + D) [1] with 64K Clear Channel response. The T1/PRI Interface PCB uses a single slot.

When installed, the T1/PRI Interface PCB uses the first block of 24 consecutive trunks. For example, if you have an 8COIU PCB installed for trunks 1-8, the T1/PRI Interface PCB will automatically use trunks 9-32. If you have 8COIU PCBs installed for trunks 1-8 and 17-24, the T1/PRI PCB will use trunks 25-48. The T1/PRI Interface cannot use trunks 9-16 (even if available) since they are not part of a consecutive block of 24 trunks.

1. Each T1/PRI Interface PCB provides 23 voice (B) and 1 data (D) channel.

Trunk Number Translation for Channel ID Based on Trunk Group Number

The Channel ID of an incoming SETUP message on a PRI line is related to the trunk group number. The system will create the trunk line number as the lowest trunk port number in the range of the same trunk group related to the channel number of the Channel ID information element of an incoming call's SETUP message.

Notes:

- In addition to T1/PRI Interface PCBs, PRI also requires a CSU/DSU Unit and interconnecting cables to interface with the telco.
- Each T1/PRI Interface PCB is switch selectable between T1 and PRI operation. For more on T1 Trunking, go to "T1 Trunking (with ANI/DNIS Compatibility)".

Two B-Channel Transfer for PRI

Depending on your software, the system may have the ability for a keyset or single line telephone to use the optional "Two B-Channel Transfer for PRI" service provided by some National ISDN carriers in the U.S. (comparable to an analog centrex).

This service supports trunk-to-trunk transfer on the telco side. With this option enabled, two trunk calls on the Aspire system can be connected together by the telco, releasing the trunks from the Aspire and connecting the outside parties together.

The Aspire can perform a trunk-to-trunk transfer in the following cases:

- Put a PRI trunk call on hold and call/answer another PRI trunk. Press the Transfer (Program 15-07 or SC 851: 06) or CONF key.
  The status and bearer capability of the two trunks both meet the service requirements.
- Put a PRI trunk call on hold and call/answer another PRI trunk, then go on-hook (Program 20-11-21 must be enabled "0").
  The status and bearer capability of the two trunks both meet the service requirements.

To use the ISDN Two B-Channel Transfer, the system sends a Facility Message to the currently active PRI trunk and then sets the retry timer (Program 24-02-09). The system will resend the Facility timer based on the retry timer. If there is no response after the 3rd resend, the system considers the transfer as failed and stops resending.

When the transfer has succeeded, the telco sends the information to the system in the Facility, Disconnect or Release Message. When the system receives a Facility Message, it sends back Disconnect or Release then releases the trunks. When the system receives a Disconnect or Release, the system simply releases the trunks (the other trunk will also receive a Disconnect or Release).

If the transfer fails, the telco sends the system Facility message with the fail information (some telcos may send an Alerting, Disconnect or Release Message). In this case, the trunks are not transferred and remain in the system.

With this feature, the following conditions apply:

- Only manual transfers (such as pressing the Transfer key or going on-hook) are supported with this feature. Aspire PRI trunks cannot perform a trunk-to-trunk transfer in the following cases:
  A. Pressing the Transfer (Program 15-07 or SC 851: 06) or CONF key while in a Conference call with two trunks (Unsupervised Conference feature).
  B. The incoming trunk call is automatically transferred to another trunk by system programming (such as Call Forward Off-Premise).
- The bearer capability of the two calls must be "Speech, 3.1-kHz Audio, Unrestricted Digital information" or compatible.

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- At least one call must have been answered. The other call must:
  - have already received ALERT message if it is an outgoing call
  OR
  - already have been answered if it is an incoming call.
- The regular Tandem Trunking programming must also be set to enable this feature.
- This feature does not support S-Bus.
- When used with Aspire Networking, this feature will work only when the two trunks are in the same system.

Basic Rate Interface (BRI)

Your system also provides compatibility with ISDN Basic Rate (BRI) services, including:

- Basic BRI Call Control (BCC)
- Point-to-Point BRI Terminal Connection (no daisy-chaining)
- Multipoint BRI Terminal Connection (daisy-chaining)
- S-Bus (allows BRI PCB's to be used as either a trunk or station interface)

BRI services require the installation of BRI Interface PCBs (2BRIU P/N 0891050 (Aspire S) or 0891006 (Aspire M/L/XL), 4BRIU P/N 0891007, 8BRIU, P/N 0891008). Each 2BRI Interface PCB has two BRI circuits. The 4BRI Interface PCB has four BRI circuits, while the 8BRI has eight BRI circuits. The BRI Interface PCBs use a single universal slot.

- **2BRI (Aspire S: P/N 0891050, Aspire M/L/XL: P/N 0891006):**
  - Provides 30 BRI circuits and 60 BRI channels.
    Each PCB takes up 8 ports. If any circuits set for T-Bus, the next set of 4 trunk ports are used. If any circuits are set up for S-Bus, the next 4 station ports are used.

- **4BRI PCB (P/N 0891007)**
  - Provides 60 BRI circuits and 120 BRI channels.
    Each PCB takes up 16 ports. If any circuits set for T-Bus, the next set of 8 trunk ports are used. If any circuits are set up for S-Bus, the next 8 station ports are used.

- **8BRI PCB (P/N 0891008)**
  - When used as T-Bus, provides 96 BRI circuits and 192 BRI channels. When used as S-Bus, 120 BRI circuits and 240 S-Bus stations ports are provided.
    Each PCB takes up 32 ports. If any circuits set for T-Bus, the next set of 16 trunk ports are used. If any circuits are set up for S-Bus, the next 16 station ports are used.

Each BRI Has Two TEI's

For each BRI line, two different TEI's will be assigned to two different SPID's.

The two different SPID's for each BRI line, will be related to different trunk logical port numbers. One BRI provides two trunk logical ports when it is connected to a CO line. Each SPID is assigned to a different TEI. This relationship is made in the initialization of the BRI line when it is connected to the CO.

This relationship between SPID and TEI's are created as follows.

LOGICAL-PORT-NUMBER + 0 = SPID-1
LOGICAL-PORT-NUMBER + 1 = SPID-2

When using the SMDR reports for BRI, all incoming BRI calls will be displayed under the CLASS column as "IVIN".

Automatic Data Link Failure Recovery

If a data link error is detected by the BRI PCB, the system will try to recover the data link and send the SPID to the central office. To provide this enhancement, the BRI PCB must be able to indicate to the system when a data link error has occurred.

Note: In addition to BRI Interface PCBs, BRI Services require the installation of NT1 Network Terminators and interconnecting cabling.

BRI and DID Callers with Non-Matching SPID Numbers

This feature allows you to determine whether the system will check the called party number with the SETUP message and the SPID setup. Depending on the system programming, this can allow DID calls to be received on BRI trunks and direct them according to the DID Translation Table (Program 22-11-03).

Ringback Tone to Telco Now Possible with BRI Trunks

An option is available using Program 10-03-17 which can be used to determine whether or not the system sends ringback tone to the telco.

Last Number Redial

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Aspire S	Aspire M/L/XL
• Available.	• Available.
• Number deletion operation enhanced with software 4.93+.	• Number deletion operation enhanced with software 4.93+.

Description

Last Number Redial allows an extension user to quickly redial the last number dialed. For example, a user may quickly recall a busy or unanswered number without manually dialing the digits.

Last Number Redial saves in system memory the last 24 digits a user dials. The number can be any combination of digits 0-9, # and *. The system remembers the digits regardless of whether the call was answered, unanswered or busy. The system normally uses the same trunk group as for the initial call. However, the extension user can preselect a specific trunk if desired.

Redial List

The system allows ***display*** telephones to have a Redial List. Up to 10 dialed numbers (both external and internal destinations) are automatically stored in the Redial List. The user can display and select one of the stored numbers and then redial it. If more than 10 destination numbers are dialed, the oldest number is automatically erased to make room for the new number.

i-Series Phones Operation Different from Aspire Phones

When an i-Series telephone is connected to the Aspire system, the Last Number Redial operation is slightly different.

When pressing the LND key, the display indicates "REDIAL [#] / ABB". The user can then press # to redial the number displayed, or enter an Abbreviated Dialing bin number to be dialed. Pressing the # key repeatedly will scroll through the last 10 numbers dialed.

Line Preference

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Line Preference determines how a keyset user places and answers calls. There are two types of Line Preference: Incoming Line Preference and Outgoing Line Preference.

Incoming Line Preference

Incoming Line Preference establishes how a keyset user answers calls. When a call rings the keyset, lifting the handset answers either the ringing call (for **Ringing Line Preference**) or seizes an idle line (for **Idle Line Preference**). The idle line can provide either Intercom or trunk dial tone (see Outgoing Line Preference below). Ringing Line Preference helps users whose primary function is to answer calls (such as a receptionist). Idle Line Preference is an aid to users whose primary function is to place calls (such as a telemarketer).

Outgoing Line Preference

Outgoing Line Preference sets how a keyset user places calls. If a keyset has Outgoing Intercom Line Preference, the user hears Intercom dial tone when they lift the handset. If a keyset has Outgoing Trunk Line Preference, the user hears trunk dial tone when they lift the handset. Outgoing Line Preference also determines what happens at extensions with Idle Line Preference. The user hears either trunk ("dial 9") or Intercom dial tone.

Auto-Answer of Non-Ringing Lines

With Auto-Answer of Non-Ringing Lines, an extension user can automatically answer trunk calls that ring other extensions (not their own). This would help a user that has to answer calls for co-workers that are away from their desks. When the user lifts the handset, they automatically answer the ringing calls based on Trunk Group Routing programming. The extension user's own ringing calls, however, always have priority over calls ringing other co-worker's extensions.

Long Conversation Cutoff

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

For incoming and outgoing central office calls, each trunk can be programmed to disconnect after a defined length of time. The timer begins when the trunk is seized and disconnects the call after the timer expires.

When used with the Warning Tone for Long Conversation feature, the system can provide a warning tone on outgoing trunks calls before the call is disconnected. This tone is not available to analog single line telephone users nor is it available for incoming calls.

Loop Keys

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Loop keys are uniquely programmed function keys that simplify placing and answering trunk calls. There are three types of loop keys: Incoming Only, Outgoing Only and Both Ways.

- **Incoming Only Loop Keys**
  Incoming Only loop keys are for answering trunk calls. An extension can have an incoming loop key for a specific trunk group (fixed) or a "catch all" loop key for any trunk group (switched). Fixed loop keys allow an extension user to tell the type of call by the ringing key. Switched loop keys are ideal for an extension with a large number of feature keys. In addition, switched loop keys are a destination for any trunk not on a line key or fixed loop key. Without a switched loop key, calls not appearing on a line key or fixed loop key will ring only the CALL key. Incoming Only loop keys also receive Transferred trunk calls.

- **Outgoing Only Loop Keys**
  Outgoing Only loop keys are for placing trunk calls. An extension can have outgoing loop keys for a specific trunk group or for ARS access. When a user presses the loop key, they get dial tone from the first available trunk in the group (or from ARS if programmed). Outgoing Only loop keys help ensure that an extension will always have a key available for placing calls.

- **Both Ways Loop Keys**
  Both Ways loop keys combine the functions of both Incoming Only and Outgoing Only loop keys. Both Ways loop keys work well for extension users that handle a moderate amount of calls and don't separate keys for incoming and outgoing calls. Both Ways loop keys also receive Transferred trunk calls.

An extension can have many loop keys - of any type. You can program an operator, for example, with four loop keys for incoming calls and four for outgoing calls.

Once a loop key call is set up, the user can handle it like any other trunk call. For example, the user can place the call on Hold, Transfer it to a co-worker or send it to a Park Orbit.

An incoming call will ring the first available loop key, beginning with the lowest numbered key. If keys 1-3 are loop keys, for example, the first incoming call rings key 1. If key 1 is busy, the next call rings key 2. If keys 1 and 2 are busy, the next call rings key 3. If all three keys are busy, additional incoming calls queue for the first available key. The telephone display will show "WAITING - LOOP KEY" if the user presses a loop key when there are additional calls waiting.

Maintenance

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Extension Data Swap function requires software 2.65+. Extension Data Swap with service code/Secure Set Relocation requires software 4.0E+.	• Extension Data Swap function requires software 2.65+. Extension Data Swap with service code/Secure Set Relocation requires software 4.0E+.
• Program 23-02-01 added to Extension Data Copy with software 2.67+.	• Program 23-02-01 added to Extension Data Copy with software 2.67+.
• Program 32-02-01 added to Extension Data Copy with software 4.93+.	• Program 32-02-01 added to Extension Data Copy with software 4.93+.
• UserPro requires software 5.93+.	• UserPro requires software 5.93+.
• Program 90-38-32 for UserPro requires software 5.96+.	• Program 90-38-32 for UserPro requires software 5.96+.

Description

The Aspire system provides several features to help maintain the Aspire database, for backup, and system analysis.

Database Maintenance
In addition to the keyset programming, the Aspire system provides the ability to use a PC to access system programming. The Windows-based PCPro and the HTML-based WebPro allows you to:

- Edit the telephone system programming options from a remote location.

  The PCPro application requires changes to be uploaded to the system before they take affect. The WebPro application applies the changes as soon as the APPLY or OK icon is clicked.

- Access system maintenance functions (like reports and tests) as well as:
  - Slot Control (possible with telephone programming or WebPro access)
    Allows the PCB slots to be deleted or reset
  - Trunk Control (possible with telephone or PCPro programming or WebPro access)
    Allows the trunks to be blocked so no new additional calls can be made on the PCB
  - Extension Control (possible with telephone programming or WebPro access)
    Allows hard or soft reset for each extension
  - System Reset (possible with telephone programming or WebPro access)
    Allows the system to be reset

In addition, PCPro allows you to:

- Save your programming to your PC's hard disk - then upload it via a LAN (Local Area Network), PPP, serial or USB connection.
  PPP is a protocol that allows a computer to use a regular telephone line and a serial interface (modem) to make TCP/IP connections.
- Download the existing programming in the telephone system via a LAN, PPP, serial or USB connection - and save it to your PC's hard disk.
- Set up a default database with the settings you use most often.
- Create a unique database for each phone system you have installed. Since you save the site-specific data to your PC's hard disk, you can easily retrieve a customer's programming if something goes wrong.

Refer to the PCPro/WebPro Installation Manual for additional details (P/N 0893201).

UserPro: End-User Programming with Web Browser Added

A programming option may be available to allow a user to adjust only the user level system programs. Similar to the WebPro application, a user would connect with an internet browser to the system. Two password levels are supported, based on the user name and password used at the sign-in, which determine the programs which can be accessed.

Mode 1 is the UA (User Programming Administrator) mode. This level allows all of the user-programmable options to be changed. The password access for this level is set up in **Program 90-02-01 : Setting the Programming Password**.

Mode 2 is the UB (User Programming) mode. This level allows a user to change only the system data pertaining to their extension. The password access for this level is set up in **Program 90-28-01 : User Programming Password Setup** and is defined for each extension user required.

The language displayed for these levels can also be changed from the English display, if required, to one of 9 other languages.

Up to 4 users can be logged in simultaneously. ***Log in is only possible when the phone is idle. The keyset for the logged in party will be placed in a locked state until logged out of user programming***.

In addition to the programs listed in **Program 90-38-xx : User Programming Data Level Setup**, Dial In Name (Program 22-11-03) is also available for access by the UA level.

UserPro Conditions

- 4 users can access the system simultaneously, but they must log in with different user accounts.
- Log in is only possible when the phone is idle. The keyset for the logged in party will be placed in a locked state until logged out of user programming.
- The displayed language is based on the system language (Program 15-02-01). If the language is changed, all the user screens are changed to the new language.
- The multiple language support is only for the User Programming feature. WebPro does not support multiple languages.
- While a user is logged in with the User Programming web browser, the system cannot be accessed by the PCPro application until the user logs out.

Extension Data Swap Function Added

Depending on your software, the system can provide the ability to swap an extension's programming to another extension number. The following extension-based programs will be swapped:

Refer to Secure Set Relocation feature below for additional options.

Program No.	Program Name
11-02	Extension Numbering
15-01	Extension Basic Data Setup (include Virtual Extension)
15-02	Multi-Line Telephone Basic Data Setup
15-03	Single Line Telephone Basic Data Setup
15-04	PHS Terminal Basic Data Setup
15-06	Trunk Access Map for Extension
15-07	Programmable Function Key
15-08	Incoming Virtual Extension Ring Tone Setup
15-09	Virtual Extension Ring Assignment
15-10	Incoming Virtual Extension Ring Tone Order Setup
15-11	Virtual Extension Delayed Ring Assignment
15-12	Conversation Recording Destination for Extension
20-06	Class of Service for Extension
21-02	Trunk Group Routing for Extensions
21-04	Toll Restriction Class for Extensions
21-07	Toll Restriction Override Password Setup
21-11	Hotline Assignment
23-02	Call Pickup Group
23-03	Ringing Line Preference
23-04	Ringing Line Preference for Virtual Extensions
24-03	Park Group Assignment
31-02	Internal Paging Group Assignment

Keep the following items in mind when using the Extension Data Swap function:

- Any user-defined programming stored in the SRAM will not be swapped (for example, Call Forward set up, Selectable Display Messaging, etc.).
- ***The extensions to be swapped must be idle*** while the swap is performed, or an "Invalid" error message will be received.
- Data for virtual extension's cannot be swapped.
- When a swap is performed, the following actions are executed for the swapped extensions.
  - Camp On Clear (Program 11-12-05)
  - Common Cancel (Program 11-12-37)
  - Last Number Redial Clear (Program 11-12-17)
  - Saved Number Clear (Program 11-12-18)
  - Incoming History data is deleted.
- ***Using Program 92-04-01 will also swap the order in which these extensions are displayed in all extension-related programs***. This means that the system will no longer display all the extension numbers from low to high. For example, if port 2 and 6 were swapped, when viewing the extensions in 15-02-01, the extensions will display in the following order: 301, 306, 303, 304, 305, 302.

Extension Data Swap by Service Code / Secure Set Relocation

The system previously provided the ability to swap an extension's programming to another extension number using **Program 92-04-01 : Extension Data Swap**. Depending on the software version, the extension data can be swapped using a service code as well.

With this option, the user must enter a 4-digit password (fixed at 4 digits) in order to complete the swap.

The following extension-based programs will be swapped:

Program No.	Program Name
11-02	Extension Numbering
12-05	Night Mode Group Assignment for Extensions
13-03	Abbreviated Dialing Group Assignment for Extensions
15-01	Extension Basic Data Setup (include Virtual Extension)
15-02	Multi-Line Telephone Basic Data Setup
15-03	Single Line Telephone Basic Data Setup
15-06	Trunk Access Map for Extension
15-07	Programmable Function Key
15-08	Incoming Virtual Extension Ring Tone Setup
15-09	Virtual Extension Ring Assignment
15-10	Incoming Virtual Extension Ring Tone Order Setup
15-11	Virtual Extension Delayed Ring Assignment
15-12	Conversation Recording Destination for Extension
15-13	Loop Keys
15-14	Programmable One-Touch Keys
16-02	Department Group Assignment for Extensions
20-06	Class of Service for Extension
21-02	Trunk Group Routing for Extensions
21-04	Toll Restriction Class for Extensions
21-07	Toll Restriction Override Password Setup
21-10	Dial Block Restriction Class Per Extensions
21-11	Hotline Assignment
21-13	ISDN Calling Party Number Setup for Extensions
21-15	Individual Trunk Group Routing for Extensions
21-18	IP Trunk Calling Party Number Setup for Extensions
21-19	IP Trunk (SIP) Calling Party Number Setup for Extensions
21-20	SIP Trunk Call Discernment Setup for Extensions
22-04	Incoming Extension Ring Group Assignment
22-06	Normal Incoming Ring Mode
23-02	Call Pickup Group
23-03	Ringing Line Preference

Program No.	Program Name
23-04	Ringing Line Preference for Virtual Extensions
24-03	Park Group Assignment
24-06	Fixed Call Forwarding
24-07	Fixed Call Forwarding Off-Premise
26-04	ARS Class of Service
26-07	Not used in U.S.
31-02	Internal Paging Group Assignment
41-02	ACD Group and Agent Assignments
41-17	ACD Login Mode Setup
42-02	Hotel/Motel Telephone Setup
92-05	Password for Extension Data Swap

Secure Set Relocation Conditions

- Any user-defined programming stored in the SRAM will not be swapped (for example, Call Forward set up, Selectable Display Messaging, etc.).
- The extensions to be swapped must be idle while the swap is performed, or an "Invalid" error message will be received.
- Data for virtual extension's cannot be swapped.
- When a swap is performed, the following actions are executed for the swapped extensions.
  - Camp On Clear (Program 11-12-05)
  - Common Cancel (Program 11-12-37)
  - Last Number Redial Clear (Program 11-12-17)
  - Saved Number Clear (Program 11-12-18)
  - Incoming History data is deleted.
- Using this option will also swap the order in which these extensions are displayed in all extension-related programs . This means that the system will no longer display all the extension numbers from low to high. For example, if port 2 and 6 were swapped, when viewing the extensions in 15-02-01, the extensions will display in the following order: 301, 306, 303, 304, 305, 302.

Extension Data Copy Function Enhanced to Provide Port Copy Option

Depending on your software, the system provides the ability to copy an extension's programming to another extension either by extension number or port number, depending on which program you use. With older software, an extension's information could only be copied based on the extension number.

The following extension-based programs can be copied:

Program No.	Program Name	Note
15-01	Extension Basic Data Setup (include Virtual Extension)	Copy all data except extension name (item 01).
15-02	Multi-Line Telephone Basic Data Setup	
15-03	Single Line Telephone Basic Data Setup	
15-04	PHS Terminal Basic Data Setup	Copy Item 11, 12 and 13.
15-06	Trunk Access Map for Extension	
15-07	Programmable Function Key	
15-08	Incoming Virtual Extension Ring Tone Setup	
15-09	Virtual Extension Ring Assignment	
15-10	Incoming Virtual Extension Ring Tone Order Setup	
15-11	Virtual Extension Delayed Ring Assignment	
15-12	Conversation Recording Destination for Extension	
20-06	Class of Service for Extension	
21-02	Trunk Group Routing for Extensions	
21-04	Toll Restriction Class for Extensions	
21-11	Hotline Assignment	
23-02	Call Pickup Group	Requires software 2.67+.
23-03	Ringing Line Preference	
23-04	Ringing Line Preference for Virtual Extensions	
24-03	Park Group Assignment	
31-02	Internal Paging Group Assignment	
32-02	Door Box Ring Assignment	Requires software 4.93+.

Keep the following items in mind when using the Extension Data Copy function:

- Using this program to copy a keyset's Programmable Function Keys will copy all the keys whether they exist on the phone to which the programming is being copied. This may cause confusion when trying to define a key which is already defined but which doesn't exist on the phone (will display as "DUPLICATE DATA"). It is recommend to either clear these non-existent keys or to only copy from an extension which has the same or fewer number of keys than the extension to which the programming is being copied.

Fill and Delete Extension Data Functions Added

The software provides the ability to fill program entries for a range of extensions to the same as a designated source extension. In addition, program data can be deleted for a range of ports.

The Fill and Delete programs can be used only with the following programs:

Program No.	Program Name
11-02	Extension Numbering
11-04	Virtual Extension Numbering
11-06	ACI Extension Numbering
11-07	Department Group Pilot Numbers
11-08	ACI Group Pilot Number
11-17	ACD Group Pilot Number

Fill and Delete Extension Data Conditions

- With the Fill function, if the data is out of range, the display will show "Invalid Data" and allow you to reenter the range.
- If data to be filled is duplicate data, the display will show "Fail to fill" and allow you to reenter the range.
- If the range of ports entered for the Delete function includes all ports, port 001 will not be deleted (to ensure phone programming can still be accessed).

Alarm Reports

The system logs various errors and information about the operation which can be used to determine the cause of a problem (up to 100 individual alarms are stored, then oldest data is deleted to allow for new information to be stored). The system can indicate the errors on a keyset's display, send the information to a printer at a programmed time, send the data via EMail. When an alarm report is printed through PCPro, the system will not delete the report data after printing. If the system is set up to EMail the report and the mail server is down, the report will not be sent.

The alarm reports indicate:

- System start-up/upgrade date and time
- PCB communication error with the date and time and the restoration date and time
- Date and time a PCB was removed from the system
- Date and time an extension was disconnected from the system
- Date and time of any interruption in system power
- Power brown-outs generate a Low Battery log
- Hard disk driver exchange time is logged (Future Item)
- Date and time of any system data change

System Information
The system can print a report of the PCBs installed, the port assignments, and the port types. This information is sent to the port defined in Program 90-13. This report includes:

- The version of system software
- PCB names
- Slot condition (working, blocked)
- Port assignment
- Port classification

Reloading Customer Databases
When reloading a customer database, the system must be reset (either using Program 90-08 or power down/power up) before all uploaded programming will take affect.

After uploading the programming, reset the system and wait a few minutes for the system to reset completely before accessing any lines or special system features. Otherwise, some unusual LED indications may be experienced.

Meet Me Conference

3

Aspire S	Aspire M/L/XL
• NTCPU provides 32 conference circuits, allowing any number of internal or external parties conferenced up to the limit of 32.	• NTCPU provides 2 blocks of 32 conference circuits, allowing each block to have any number of internal or external parties conferenced up to the block's limit of 32.

Description

With Meet Me Conference, an extension user can set up a Conference with their current call and up to 32 other internal or external parties. Each party joins the Conference by dialing a Meet Me Conference code. Meet Me Conference lets extension users have a telephone meeting -- without leaving the office.

Meet Me Paging

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Meet Me Paging allows an extension user to Page a co-worker and privately meet with them on a Page zone. The Paging zone is busy to other users while the meeting takes place. While the co-workers meet on the zone, no one else can hear the conversation, join in or make an announcement using that zone. Meet Me Paging is a good way to talk to a co-worker when their location is unknown. If the co-worker can hear the Page, they can join in the conversation.

Meet Me Paging Transfer

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

If a user wants to Transfer a call to a co-worker but they don't know where the co-worker is, they can use Meet Me Paging Transfer. With Meet Me Paging Transfer, the user can Page the co-worker and have the call automatically Transfer when the co-worker answers the Page. Since Meet Me Paging Transfer works with both Internal and External Paging, a call can be quickly extended to a co-worker anywhere in the facility.

Memo Dial

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

While on an outside call, Memo Dial lets a display keyset user store an important number for easy redialing later on. The telephone can be like a notepad. For example, a user could dial Directory Assistance and ask for a client's telephone number. When Directory Assistance plays back the requested number, the caller can use Memo Dial to jot the number down in the telephone's memory. They can quickly call the Memo Dial number after hanging up.

When a user enters a Memo Dial number, the dialed digits do not output over the trunk. Dialing Memo Dial digits does not interfere with a call in progress.

Message Waiting

Aspire S	Aspire M/L/XL
• Available.	• Available.
• The MSG key as Voice Mail key feature is available.	• The MSG key as Voice Mail key feature is available with 1.11+ software.
• LED Color Indication option requires software 2.63+.	• LED Color Indication option requires software 2.63+.
• SLT MW Indication option requires software 2.63+.	• SLT MW Indication option requires software 2.63+.

Description

An extension user can leave a Message Waiting indication at a busy or unanswered extension requesting a return call. The indication is a flashing MW lamp at the called extension and a steadily lit MW lamp on the calling extension. Answering the Message Waiting automatically calls the extension which left the indication. Message Waiting ensures that a user will not have to recall an unanswered extension. It also ensures that a user will not miss calls when their extension is busy or unattended. Additionally, Message Waiting lets extension users:

- View and selectively answer messages left at their extension (display keyset only)
- Cancel all messages left at their extension
- Cancel messages they left at other extensions

An extension user can leave Messages Waiting at any number of extensions. Also, any number of extensions can leave a Message Waiting at the same extension. A periodic VRS announcement may remind users that they have Messages Waiting. Up to 50 Message Waiting indications can be displayed in a system at one time.

MSG Key will Operate as Voice Mail Key

The system enhances a phone's MSG key function when connected to a system which has voice mail installed. When an extension receives a voice mail, the MSG key can be used to check the number of messages in voice mail, as well as call the voice mail to listen to the messages. If there is no Voice Mail Programmable Function Key defined (Program 15-07-01, code 77), the phone's Message Waiting LED will flash to indicate new messages.

This option is not available with a networked voice mail - the voice mail must be local.

Refer to the Voice Mail feature for the feature operation.

LED Color Indication

Depending on your software, the software provides an option allows you to select whether the Message Wait LED located at the top of the keyset will flash green (0) or red (1) when a Message Wait indication is flashing. By default, this option is set to flash red.

Note that if this LED is also used for voice mail indications (no Programmable Function Key programmed for voice mail), if there are both voice mail messages and Message Wait indications, the color set for Message Wait will override the color used for voice mail indications (red).

Single Line Telephone MW Indication Option

An option is available for analog single line telephones which provide a display. When a user leaves a Message Waiting for a SLT which has a display, **Program 15-03-13 : Single Line Telephone Basic Data Setup - MW Signal Type** is used to determine whether the SLT user will see a MW LED indication or if the Caller ID will be used to display the call.

Microphone Cutoff

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Microphone Cutoff lets a keyset user turn off their phone's handsfree or handset microphone at any time. When activated, Microphone Mute prevents the caller from hearing conversations in the user's work area. The user may turn off the microphone while their telephone is idle, busy on a call or ringing. The microphone stays off until the user turns it back on.

Multiple Directory Numbers / Call Coverage

Aspire S	Aspire M/L/XL
• Available - Virtual ports 1-24.	• Available - Virtual ports 1-256.
• Virtual extensions can use Fixed Call Forward Off-Premise with CO trunks with software 2.63+.	• Virtual extensions can use Fixed Call Forward Off-Premise with CO trunks with software 2.63+.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.
• Option to change the virtual key lamping on a per extension basis requires software 5.91+.	• Option to change the virtual key lamping on a per extension basis requires software 5.91+.

Description

Multiple Directory Numbers let a keyset have more than one extension number. Calls can route to the keyset's installed number or to the keyset's "virtual extension" Multiple Directory Number key. This helps users identify incoming calls. For example, an extension installed at 304 (Sales) could have a virtual extension for 460 (Service). Calls to 304 ring the extension normally. Calls to 460 ring the Multiple Directory Number key. This lets the user at extension 304 differentiate Sales calls from Service calls. After answering a call, based on extension or system-defined options, the call can remain on the Multiple Directory Number key or it can switch to a CALL key, line key or loop key.

Call Coverage

A keyset can have Multiple Directory Number keys set up as Call Coverage keys for co-worker's extensions. The Call Coverage key lights when the co-worker's extension is busy and flashes slowly when the co-worker has an incoming call. The Call Coverage key can ring immediately when a call comes into the covered extension, ring after a delay or not ring at all. In addition, the keyset user can press the Call Coverage key to intercept their co-worker's incoming call. The user can also go off hook and press the Call Coverage key to call the covered extension.

If the covered extension is busy and they receive a second call, the covering extension's Call Coverage key will flash. The user just presses the flashing key to pick up the call.

The Call Coverage keys follow the extension's Do Not Disturb and Off-Hook Signaling programming. These keys do not, however, indicate the lamping for extensions in DND. If this is required, a Hotline key can be used instead.

Place and Receive Calls on Call Coverage/Multiple Directory Number Keys

Multiple Directory Number keys/Call Coverage keys can be used three separate ways, depending on how the key is defined in Program 15-02-21.

- a DSS key to the extension and for receiving incoming calls
- answering incoming calls with the ability to place outgoing ICM or CO calls
  OR
- just for receiving incoming calls

A keyset can have Multiple Directory Number/Call Coverage keys for many different extensions and virtual extensions. In addition, co-workers can share the same Multiple Directory Numbers. For example, everyone in the Service Department could have a key for the Sales Department's virtual extension.

Auto Off-Hook Answer and Ringing Line Preference for Call Coverage Keys

An extension's Call Coverage Keys can be programmed to allow the user to simply pick up the handset to answer a ringing call. So as not to interfere with ringing trunk or Intercom calls, the system automatically assigns Call Coverage Key ringing with the lowest answering priority. If multiple Call Coverage Keys are ringing, answering priority is set first by the assigned ring pattern and then by the key position.

Virtual Extension vs. Ring Groups

As the system does not allow voice mail calls to ring Ring Groups, a virtual extension can be created which will allow you to direct more than one call to the extension. The system will allow up to 10 calls to be queued. When you program a Call Coverage Key for that extension to ring, the next call can then be answered.

This could allow a voice mail caller to dial the virtual extension and have all the extensions with the same virtual extension key ring. The system can be programmed as follows:

- Program 11-04, 15-01-01: Assign a virtual extension number and name (example: virtual port 1, extension 5400, Sales, virtual port 2, extension 5401, Customer Service, etc.).
- Program 11-07: Assign a Department Group Pilot number for the virtual extension (example: Department Group Number 2 = 5555).
- Program 15-07: Assign a Call Coverage key (*03) to an extension for the virtual extension number assigned.
- Program 16-02: Department Group Assignment for Virtual Extensions (example: virtual extension 5400 - group 2, virtual extension 5401 - group 2).

The end user can then simply transfer the call to the virtual extension number (example: 5555). The call is in placed in queue and will be answered in turn as soon as the extension is available.

Call Forward Off-Premise From CO Trunk/Voice Mail Transfer Possible

The system allows virtual extensions to use Fixed Call Forward Off-Premise with normal central office trunks. With some older software, this was only possible using DIL or DID trunks. This enhancement allows a call transferred by the voice mail to a virtual extension to be forwarded off-premise.

Music on Hold

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Using a system tone for MOH requires software 2.63+.	• Using a system tone for MOH requires software 2.63+.
• Using VRS for MOH requires software 5.15+.	• Using VRS for MOH requires software 5.15+.

Description

Music on Hold (MOH) sends music to calls on Hold and parked calls. The music lets the caller know that his call is waiting, not forgotten. Without Music on Hold, the system provides silence to these types of calls. The Music on Hold source can be internal (synthesized) or from a customer-provided music source (i.e., tape deck, receiver, etc.). The customer-provided source can connect to a PGDAD module analog port or to a connector on the CPRU PCB.

Note: In accordance with U.S. copyright law, a license may be required from the American Society of Composers, Authors and Publishers (ASCAP) or other similar organizations, if radio, television broadcasts or music other than material not in the public domain are transmitted through the Music on Hold feature of telecommunications systems. NEC America hereby disclaims any liability arising out of the failure to obtain such a license.

Option Available for Using System Tone

The Music on Hold feature has been enhanced to allow callers to hears a system tone instead of playing the internal or external music.

VRS for Music on Hold

A user can now save their own music to the DSPDBU VRS to use as the Music on Hold source.

When using the VRS for Music on Hold, the music file must be CCITT μ-law 8kHz 8-Bit Monaural PCM data. This type of file can be easily encoded using the Windows Sound Recorder application or similar software). Any other type of formatted music file will be played as silence.

The music file must be saved in the **\VM\OGM\3\1\9** directory of the VRS compact flash card and it must be named “**Gxx.WAV**”. The “xx” can be a two-digit number from 00 to 47, which is associated with Program 10-04-02 by subtracting 1.

This “xx” is also associated with the regular VRS message number (01 to 48), so the user can record the music via phone by using the service code 116 + 7 instead of using the PC.

VRS for MOH Conditions

- When the IntraMail compact flash is installed, this feature cannot be used.
- When the selected music file is not accessible (for example, the file does not exist, the DSPDBU is not installed, etc.), or all the VRS channels are busy, the system will play Internal MOH (Selection 1) instead.

Name Storing

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Additional characters available.	• Additional characters added in software 2.05+.

Description

Extensions and trunks can have names instead of just circuit numbers. These names show on a key-set's display when the user places or answers calls. Extension and trunk names make it easier to identify callers. The user does not have to refer to a directory when processing calls. A name can be up to 12 digits long, consisting of alphanumeric characters, punctuation marks and spaces.

Additional Characters Available

When using the Name Storing feature, the system now provides additional characters which can be used. These characters are available with any option which allows Name Storing - Abbreviated Dialing, One-Touch Keys, Extension Name, Trunk Naming.

Under key 1:

Á À Â Ã Ç É Ê ì ó

Under key 0:

ô Õ ú ä ö ü a e θ

Under key *:

π ∑ σ Ω ∞ ¢ £

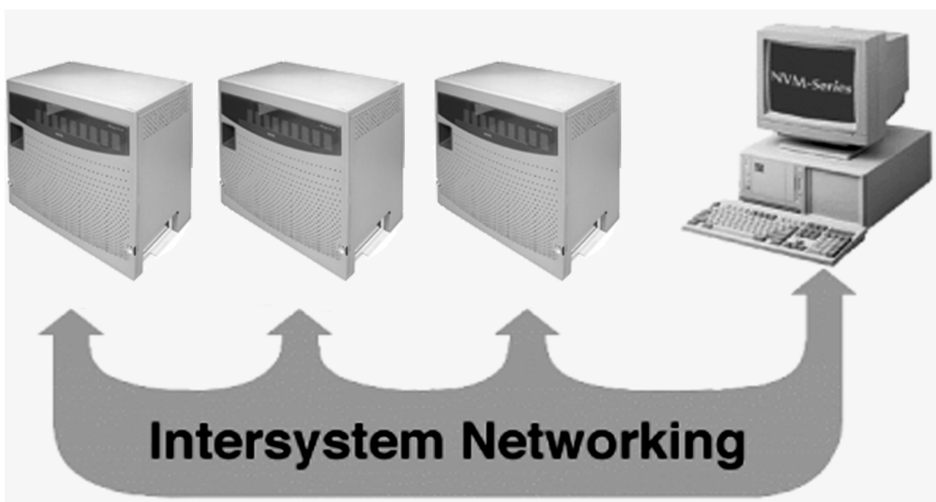
With this change, to enter a space or accept an entry, the # key is used. However, this key will only work when performing user programming (such as Name Storing for an extension, service code 800). When in system programming, the right arrow soft key must be used to advance the cursor.

3

Networking

Aspire S	Aspire M/L/XL
• Networking with VoIP or BRI available. PRI Networking is not available.	• Two System Networking available with software 1.07+. • Multiple Site Networking available with software 2.08+. • This feature requires the Basic NTCPU (P/N 0891002) with the PAL upgrade or the Enhanced NTCPU (P/N 0891038).
• Keep Alive options added with software 1.11+.	• Keep Alive options added with software 1.11+.
• PRI Channel Limitation is not available.	• PRI Channel Limitation added with software 2.09+.
• Transfer Network trunk to local voice mail using voice mail key requires software 3.05+.	• Transfer Network trunk to local voice mail using voice mail key requires software 3.05+.
• Enhanced Fax Over Networking requires software 4.93+.	• Enhanced Fax Over Networking requires software 4.93+.

Description



Use the built in networking feature to integrate multiple phone systems into a single "virtual" communications system. Interconnected with ISDN PRI or BRI or VoIP, each phone system becomes a node on the network that can communicate with any other phone system node.

- **Centralized Network Attendant**
  Centralized Network Attendant allows multiple networked systems to share a single centralized attendant. This centralized attendant can receive calls from and transfer calls to any destination in any network node. Unanswered calls recall and route as if they were part of a single, much larger system.

- **Flexible Network Routing**
  Use network routes to set up ISDN and VoIP networking between many separate systems - or use mixed (ISDN or VoIP) networking per system for greater network performance. Data tables in the system program define the routing for each network node. These tables are easily customized to meet the requirements of each networking configuration.

  Users may place an intercom call or transfer a call to any extension at any location by simply dialing an extension number. The system analyzes each extension number received and determines how to route the call to its final destination. The feature which handles this route selection is called Flexible Routing (F-Routing). F-Routing also has the ability to select alternate routes to the destination extension if the primary destination is busy. Up to 48 routes are available for networking. Once an extension number is dialed, the system checks the routing, accesses the assigned trunk group and places the call. Each extension is assigned a route or routes that decides which trunk group to access and any modified dialed data if required.

  When an Aspire system is connected via a VoIP connection to an i-Series system in a tie-line type setup, in order to transfer calls from the Aspire system to the i-Series, in addition to the VoIP programs specified in the VoIP feature (page 221), set up the Flexible Routing Tables as follows:

  - **Program 44-05-01 : ARS/F-Route Table** ; Table Number 1 = 9 (Trunk Group for Aspire VoIP Trunk)
  - **Program 10-23-02 : H.323 System Interconnection, IP Address** ; System Number 3 = 172.16.9.10 (IP Address for i-Series System)
  - **Program 10-23-04 : H.323 System Interconnection, Alias Address** ; System Number 3 = 4 (For Dial 4 Calls)

  With this programming, the Aspire system will wait for the Trunk Interdigit Timer to expire before dialing out after an i-Series extension (4xxx) is dialed.

  If the F-Routing is set up with Program 44-05-01; Table Number 1 set to 103 (Networking) and Program 10-27-01; System ID 3 = 172.16.9.10 (IP Address for i-Series), though the i-Series system will be able to transfer calls to the Aspire, the Aspire system will not be able to transfer to the i -Series.

PRI Networking With Two Local Voice Mails, Masters Must Have Different Numbering

When programming a PRI network with each system having their own local voice mail, the master numbers for the voice mails must be defined in different series in Program 11-01. The second digit of the extension number can not be the same. For example, 700 and 701 will not work, however 700 and 710 can be used.

With Two Local Voice Mails, Network ID Must Match in Programming

With a networked system, with each system having their own voice mail. The system must be programmed correctly in order for users to properly connect to the correct voice mail when leaving a message. Refer to the Aspire Software Manual for details.

Multiple Site Networking Available

IP Networking allows for a maximum of 50 nodes. With ISDN Networking, the maximum number of nodes depends on the type of trunks used - PRI, BRI or a mixture of both. As the Aspire M/L is limited to 8 PRI PCBs, when PRI trunks are used, the limit would be 8 systems, but each system would require 8 PRI PCBs. ***PRI Networking is not available with the Aspire S.***

The recommended connection for multiple system Networking is to interconnect all the systems using hardware (required with ISDN Networking) and programming. This also allows such features as Park to be used in a network.

With VoIP Networking, the system IDs for each networked system are defined in **Program 10-27 : IP System ID**. An IP address is defined for each node, and using the Numbering Plan (Program 11-01xx), the dialed digits are analyzed, the system ID is determined, and the call is routed to the networked system.

The networked systems can be interconnected using a combination of PRI and VoIP trunks. In this type of setup, the IP resources will received priority over the PRI resources. The PRI will be used when the IP resources are busy.

With ISDN Networking, interconnecting the systems is accomplished by defining a master PCB and slave PCB between each system (**Program 10-03-10 : PCB Setup - Master/Slave System**). Each system must have an ISDN PCB for each other system in the network.

The networked systems can be interconnected using a combination of PRI and VoIP trunks. In this type of setup, the IP resources will received priority over the PRI resources. The PRI will be used when the IP resources are busy. The following two graphics indicate correct networking layouts. Each one provides connection between all three systems within the network.

ISDN Networking Clock Source

When using ISDN Networking (PRI or BRI) on the Aspire system, the option selected in **Program 10-03-01 : PCB Setup - ISDN Line Mode** determines the clock source for the networked connection. The following information indicates how, with each option, the clock source is obtained for the networked systems.

<u>Option 3: Network Mode (Leased Line)</u>
Telco sends the clock to the Master System
Telco sends the clock to the Slave System

<u>Option 4: Network Mode (Interconnected Line)</u>
Master System sends the clock to the Telco (or direct connection without telco) which then sends the clock to the Slave System

<u>Option 5: (Interconnected Line, Fixed Layer 1=NT)</u>
Master System sends the clock to the Telco
Slave System sends the clock to the Telco

Keep Alive Programs Added

Two options in Program 10-31 for use with the Networking feature are available to define the interval of Keep Alive and how many times the system resends Keep Alive.

PRI Channel Limitation

The system provides an option which can be used to assign the number of B-channels to be used for each ISDN PCB. This allows for fractional PRIs when used with multiple site networking. If this program is limited to less than "23" on one side of the network, then it also limits both inbound and outbound network calls. This also applies on the other side of the network as well.

Transfer Network Trunk to Local Voice Mail Using Voice Mail Key

With older software, IP or PRI network sites that had their own voice mail could not transfer a call into voice mail using the following steps if the inbound call originated in another site (for example, a call comes in to Site A and it translates to an extension at Site B):

- HOLD + the Voice Mail Programmable Function Key (Program 15-07 or SC 851 + 77) or MSG key (Program 15-02-26=1) + the extension number.

Using software 3.05 or higher, this operation is now possible. Keep the following in mind when using this option:

- ***Note that if you have a local and central voice mail, you can not have the same mailbox number at each node.*** If both sites have the same mailbox, when transferring a call, it will be transferred into the local voice mail and not across the network.
- As this software change allows transferring to a general message by a Single Line Set, the following two operations have different results between Networking Call Transfer and Internal Call Transfer.
  - Hold + Voice Mail Master Dial then hang up.
  - Hold + Service Code (Own Mailbox Access - *8) then hang up.

Fax Over Networking

The purpose of Aspire Networking is to be able to connect several systems and have them appear to operate as one system. However, some restrictions still apply. With older software, with Fax Over Networking using H.323 trunks, if a resource was busy, the operation could not be performed efficiently. Although the operation would continue if there were no G.711 compression, there was no resending procedure with RTP and reliability was a problem. The software now enhances this operation to provide better performance.

With IP networking, the modem signal of the fax relay now uses H.245. This enhancement only applies to G3.

For additional information on Networking, refer to the Aspire Networking Guide (P/N 0893207).

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Night Service

Aspire S	Aspire M/L/XL
• Available - 4 Night Service Patterns/Groups.	• Available - 32 Night Service Patterns/Groups.
• Toggle night modes with Programmable Function Key requires software 2.63+.	• Toggle night modes with Programmable Function Key requires software 2.63+.

Description

Night Service lets system users activate one of the Night Service modes. Night Service redirects calls to their night mode destination, as determined by Assigned and Universal Night Answer programming. The system allows Night Service to be enabled for each Night Service group, allowing each group to determine when their calls should switch modes. A user typically activates Night Service after normal working hours, when most employees are unavailable to answer calls. The system also provides external contacts to enable Night Service.

There are four (Aspire S) or eight (Aspire M/L/XL) Night Service modes:

- Day 1 / Day 2 Modes - for normal working hours
- Night 1 / Night 2 Modes - after hours (usually evening)
- Midnight 1 / Midnight 2 Modes - late at night to early in the morning
- Rest 1 / Rest 2 Modes - interval usually used for lunch

Assigned Night Answer (ANA)
With Assigned Night Answer, Night Service has calls ring extensions directly. Assigned Night Answer provides an answering point for Night Service calls. For certain applications, this may be more appropriate than Universal Night Answer. For example, you could program trunks to ring the security station telephone during off hours.

Universal Night Answer (UNA)
Universal Night Answer makes incoming calls ring over the External Paging speakers. With UNA, an employee can go to a telephone and press the flashing line key or use “Universal Answer” to pick up the call. Only ring groups calls can be used with Universal Night Answer. For more on setting up Universal Answer, turn to the “Central Office Calls, Answering” feature.

You may also be able to use Transfer to UNA. An extension user can Transfer their call to UNA (i.e., External Paging at night). Once transferred, the call will ring the External Paging speakers like any other UNA call and can be picked up at any extension. You can also set up Transfer to UNA through the VRS. This lets outside callers, answered by the VRS, dial a code to have their call ring External Paging.

Automatic Night Service
The system will allow or deny Automatic Night Service based on the extension’s class of service programming. If allowed, the calls will then route according to the service patterns programmed. The Night Service programming is stored in the RAM memory. This means that if the system is not using the Automatic Night Service, in the case of a power failure while in night mode, when the power is restored, the system will continue to be in night mode.

Programmable Function Key Can Toggle Night Modes
The software allows a Night Service Programmable Function Key (PGM 15-07 or SC 851: 09 + 0) to toggle night modes. You can determine in programming how many modes through which the user will toggle. Note that the additional data for the Programmable Function Key must be set to "0" for the toggle function to work.

Off Hook Signaling

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

When a user calls an extension busy on a call, they can send an off hook signal through the handset and through the telephone’s speaker indicating they are trying to get through. The signal is an off hook ringing over the idle (second) line appearance. Off Hook Signaling helps important callers get through, without waiting in line for the called extension to become free. The system provides the following Off Hook Signaling options:

- **Called Extension Block**
  The called extension’s Class of Service may block incoming Off Hook Signaling attempts. This is beneficial to users that don’t want interruptions while on a call.
- **Automatic Signaling**
  Calling a busy extension automatically initiates Off Hook Signaling. This option is useful to receptionists, operators and others that must quickly process calls. This is set in the called extension’s Class of Service.

- **Manual Signaling**
After reaching a busy extension, manual signaling gives the caller the choice of using Off Hook Signaling or activating other features. Extension's without automatic signaling have manual signaling. The users can dial a service code or press a Programmable Function Key to send Off Hook Signaling to the called phone.
- **Selectable Off Hook Signaling Mode**
The Off Hook Signal can be muted ringing, no off hook ringing or a beep in the handset - based on the caller's programming.
- **Off Hook Ringing**
Use this option to enable or disable an extension's Off Hook Signaling for incoming calls. If enabled, Off Hook Signaling occurs normally. If disabled, calls queue behind the extension's busy line appearance and the user gets no Off Hook Signaling indication. The second line appearance stays idle. The caller hears ringback tone while their call waits. This is set in the called extension's Class of Service.
- **DID Call Waiting**
An extension can optionally have DID calls camp on with Off Hook/Call Wait signaling, without Off Hook/Call Wait signaling or no signaling. This is set in the called extension's Class of Service.
- **Block Manual Off Hook Signals**
This Class of Service option enables/disables a busy extension's ability to block off hook signals manually sent from a co-worker. If disabled (not blocked), callers can dial 7 at busy or busy/ring to signal the extension. If enabled (blocked), nothing happens when the caller dials 7 to off hook signal.
- **Block Camp On**
If an extension has Block Camp On enabled, callers to the extension cannot dial 2 to Camp On after hearing busy or busy/ring. If the extension has Block Camp On disabled, callers are not prevented from dialing 2 to Camp on after hearing busy or busy/ring. This is set in the called extension's Class of Service.

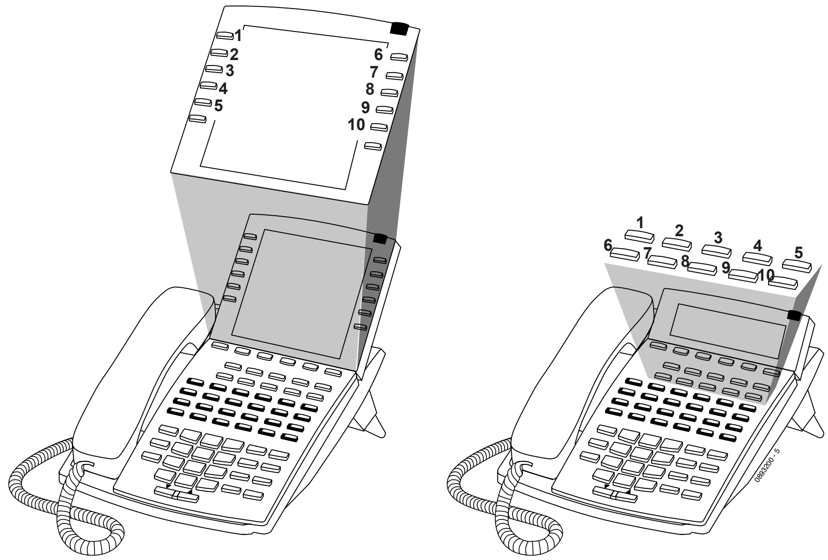
One-Touch Calling

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

One-Touch Calling gives a keyset user one button access to extensions, trunks and selected system features. This saves users time when accessing co-workers, clients and features they use most often. Instead of dialing a series of codes, the user need only press the One-Touch Key. An extension user can have One-Touch Keys programmed for:

- **Direct Station Selection** - one button access to extensions
- **Personal Speed Dial** - one button access to stored numbers (up to 24 digits long)
- **Abbreviated Dialing** - one button access to stored Abbreviated Dialing numbers
- **Trunk Calling** - one button access to trunks or trunk groups
- **Service Codes** - one button access to specific Service Codes



An extension user can chain dial with One-Touch Keys. For example, a user can store the number for a company's Automated Attendant in key 1 and employee extension numbers in keys 2-5. The user presses key 1 to call the company, then one of keys 2-5 to ring the employee to which they want to speak.

An extension user or system administrator can optionally store a Flash command under a One-Touch Key. This is helpful for One-Touch Keys used as Personal Speed Dial bins. The stored Flash may be helpful to access features of the connected telco, PBX or Centrex.

Operator

Aspire S	Aspire M/L/XL
• Available - up to 8 operators.	• Available - up to 8 operators.

Description

When an extension user dials "0", calls are routed to a main system operator. The operator can answer and route outside calls or locate employees using the Page feature.

OPX (Off Premise Extension)

Refer to the **Single Line Telephones, Analog 500/2500 Sets** (page 186) for information on this feature.

Paging, External

Aspire S	Aspire M/L/XL
• Available - 8 External Paging zones.	• Available - 9 External Paging zones.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.

Description

With External Paging, a user can broadcast announcements over paging equipment connected to external Paging zones. When a user pages on of these external zones, the system broadcasts the announcement over the speakers. Like Internal Paging, External Paging allows a user to locate another employee or make an announcement without calling each extension individually.

The Aspire S system allows up to eight External Paging zones. The Aspire M/L/XL system allows up to nine External Paging zones, with the additional zone (#9) provided on the NTCPU. All other zones (#1-8) requires a port on a 2PGDAD module, with a maximum of two external paging circuits per module. You must have four 2PGDAD modules to get the eight external zones. In addition, each external zone has an associated relay contact. When a user pages to a zone, the corresponding contact activates (closes). This provides for Paging amplifier control. Refer to the system hardware manual for additional details.

Combined Paging

Use Combined Paging when you want to simultaneously Page into an internal and corresponding external zone. For example, you can Page your company's warehouse and outside loading dock at the same time. Combined Paging is available for zones 1-8 and All Call. Refer to **Paging, Internal** (page 174) for more on setting up Combined Paging. In addition, you can program a Function Key as a Combined Paging key. Using the External Page Function Key, when an All Call External Page Function Key is programmed, it will include both the external zones and the assigned internal zone(s). If the internal page zone is busy or there are no extensions in a page group, the announcement will be made on the external zones only.

Remove Paging Information from Display Phones

A Class of Service option is available in system programming to prevent display telephones from showing incoming paging information. This allows the system to save processor time and speed up system operation.

Paging, Internal

Aspire S	Aspire M/L/XL
• Available - 8 Internal Paging Groups (Zones).	• Available - 64 Internal Paging Groups (Zones).
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.

Description

Internal Paging lets extension users broadcast announcements to other keyset users. When a user makes a Zone Paging announcement, the announcement broadcasts to all idle extensions in the zone dialed. With All Call Paging, the announcement broadcasts to all idle extensions programmed to receive All Call Paging. An extension can be a member of only one Internal Paging Zone. Like External Paging, Internal Paging allows a user to locate another employee or make an announcement without calling each extension individually.

Combined Paging

Use Combined Paging when you want to simultaneously Page into an internal and corresponding external zone. For example, you can Page your company's warehouse and outside loading dock at the same time. Combined Paging is available for Paging zones 1-8 and All Call. Optionally, you can change the Combined Paging assignments. For example, you can associate External Paging Zone 1 with Internal Paging Zone 4. You can program a Function Key as a Combined Paging key. When an All Call External Page Function Key is programmed, it will include both the external zones and the assigned internal zone(s). If the internal page zone is busy or there are no extensions in a page group, the announcement will be made on the external zones only.

Remove Paging Information from Display Phones

A Class of Service option is available in system programming to prevent display telephones from showing incoming internal paging information. This allows the system to save processor time and speed up system operation.

Paging, Privacy Release

Please refer to **Conference, Voice Call/Privacy Release** (page 111) for information on this feature.

Park

Aspire S	Aspire M/L/XL
• Available - 64 System Park orbits.	• Available - 64 System Park orbits.
• Personal Park Programmable Function Key and Service Code requires software 2.65+.	• Personal Park Programmable Function Key and Service Code requires software 2.65+.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.
• Personal Park at a co-worker's extension requires software 5.91+.	• Personal Park at a co-worker's extension requires software 5.91+.

Description

Park places a call in a waiting state (called a Park Orbit) so that an extension user may pick it up. There are two types of Park: System and Personal. Use System Park when you want to have the call wait in a system orbit. Personal Park allows a user to Park a call at their extension so a co-worker can pick it up. After parking a call in orbit, a user can Page the person receiving the call and hang up. The paged party dials a code or presses a programmed Park key to pick up the call. With Park, it is not necessary to locate a person to handle their calls. A call parked for too long will recall the extension that initially parked it, however the call remains in the park orbit until it's answered. There are 64 Park Orbits (1-64) available for use.

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If Caller ID is enabled in the system, a user can check the Caller ID for a parked call using their Park key and possibly the Flash key (depending on programming).

Splitting Between Parked Calls

A keyset user can retrieve two calls from Park Orbit (for which they don't have line appearances) and easily split (alternate) between them. The split operation brings the calls to the user's telephone and frees up the Park Orbits.

Extended Park

An extension's Class of Service determines whether it will use the normal Park Orbit Recall time or the Extended Park Orbit Recall time. The timers are set up in system programming. When an extension with Extended Park Recall Class of Service option parks a call, it recalls after the Extended Park Orbit Recall time. When an extension with the Normal Park Orbit Recall Class of Service option parks a call, it recalls after the normal Park Orbit Recall time, however the call remains in the park orbit until it's answered.

Programmable Function Key and Service Code Available for Personal Park

The Personal Park feature is enhanced with the ability to use a Programmable Function Key or service code (3-digit or 1-digit) to place a call in Personal Park. Older software only provides the option to place a call in Personal Park using a 3-digit service code. With this ability, two new program options have been added to the system in Program 11-16-11 and 15-07-01. This option is available for keysets, single line sets, and Aspire Wireless telephones and can be used for analog or ISDN trunks.

With this feature, the following conditions apply:

- An extension can have only one Personal Park key.
- When the terminal that has a call in Personal Park is unplugged, the Personal Park will be released and the held caller will receive busy tone.

The following table indicates what condition the service codes and Programmable Function key can be used.

Status	Using 3-Digit Service Code	Using 1-Digit Service Code	Using Personal Park Key
Speaking	Not Available	Not Available	Available
ICM Dial Tone or Busy Tone	Available	Not Available	Available
Calling Another Extension	Available	Available (with outside call on hold and when called extension does not answer)	Available
Receiving a Personal Park Recall	Available	Not Available	Available

Personal Park at a Co-Worker's Extension
The Personal Park feature is enhanced to allow an extension user to place an outside call, which is on hold, on Personal Park at a co-worker's extension after placing an intercom call. This feature is available for keysets, SLTs, IP phones and Aspire Wireless phones.

PBX Compatibility

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

You can connect your phone system trunks to Centrex/PBX lines, rather than to telco trunk circuits. This makes the trunk inputs into the system 500/2500 type compatible Centrex/PBX extensions, rather than telco circuits. PBX Compatibility lets the system be a node (i.e., satellite) in a larger private telephone network. To place outside calls when the system is behind a PBX, phone system users must first dial the PBX's trunk access code (usually 9).

The system provides the following PBX Compatibility options:

- **PBX Trunk Access Code Screening**
  The system can monitor the numbers users dial and screen for PBX trunk access codes. The system can screen for up to 4 groups of trunk access codes. The codes can be one or two digits long, consisting of the digits 0-9, # and *. (You use Line Key 1 as a wild card entry.)

- **PBX Trunk Toll Restriction**
  The system can provide the Toll Restriction for the PBX trunk, or restriction can be handled solely by the connected PBX. If the phone system provides the restriction, it restricts the digits dialed after the PBX access code.

- **PBX Call Restriction**
  When the phone system does the Toll Restriction, it can further restrict users from dialing PBX extensions. In this case, the only valid numbers are those dialed after the PBX trunk access code. The only PBX facility phone system users can access are the PBX's outside trunks.

- **Automatic Pause**
  The system automatically pauses when it sees a PBX trunk access code during manual dialing, Abbreviated Dialing, Last Number Redial, Repeat Redial and Save Number Dialed. This gives the connected PBX time to set up its trunk circuits.

Prime Line Selection

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Prime Line Selection allows a keyset user to place or answer a call over a specific trunk by just lifting the handset. The user does not have to first press keys or dial codes. This simplifies handling calls on a frequently used trunk.

Prime Line Selection has the following two modes of operation:

- **Outgoing Prime Line Preference**
  Lifting the handset seizes the Prime Line. Outgoing Prime Line Preference would help a telemarketer who always needs a free line to call prospective clients. The telemarketer just lifts the handset and the Prime Line is always available. (Outgoing Prime Line Preference may be affected by Incoming Prime Line Preference -- see Programming below.)

- **Incoming Prime Line Preference**
  When the Prime Line rings the extension, lifting the handset answers the call. Incoming Prime Line Preference could benefit the Service Department dispatcher who must quickly answer customer's service calls and then dispatch repair technicians. The dispatcher would have the assurance than whenever a customer calls in, the dispatcher just lifts the handset get their call. (Incoming Prime Line Preference can optionally seize an idle line appearance -- see Programming below.)

Privacy Release

Please refer to Conference, Voice Call/Privacy Release (page 278) **for information on this feature.**

Private Line

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

A Private Line is a trunk reserved for a keyset for placing and answering calls. A user with a Private Line always knows when important calls are for them. Additionally, the user has their own trunk for placing calls that is not available to others in the system.

- **Incoming only**
  The keyset has a Private Line only for incoming calls. The user cannot place calls on the Private Line.

- **Outgoing only**
  The keyset has a Private Line only for outgoing calls. The Private Line does not ring for incoming calls.

- **Both ways**
  The keyset has a Private Line for both incoming and outgoing calls.

Programmable Function Keys

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Each keyset has Programmable Function Keys. Programmable Function Keys simplify placing calls, answering calls and using certain features. You can customize the function of a keyset's programmable keys from your administration telephone, or the extension user can do it themselves. Depending on your telephone style, you can have either 12 or 24 Programmable Function Keys.

Pulse to Tone Conversion

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

An extension can use Pulse to Tone Conversion on trunk calls. Pulse to Tone Conversion lets a user change their extension's dialing mode while placing a call. For systems in a Dial Pulse area, this permits users to access dial-up OCCs (such as MCI) from their DP area. The user can, for example:

- Place a call to an OCC over a DP trunk.
- Depending on programming:
  Manually implement Pulse to Tone Conversion
  OR
  Wait 10 seconds.
- Dial the OCC security code and desired number. The system dials the digits after the conversion as DTMF.

3

Repeat Redial

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

If a keyset user places a trunk call that is busy or unanswered, they can have Repeat Redial try it again later on. The user doesn't continually have to try the number again -- hoping it will go through. Repeat Redial automatically retries it until the called party answers (the number of retries is based on system programming).

Reverse Voice Over

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

While on a handset call, Reverse Voice Over lets a busy keyset user make a private Intercom call to an idle co-worker. The idle co-worker can be at a keyset or 500/2500 set. The busy user just presses and holds down a programmed Reverse Voice Over key to make a private call to a specified co-worker. The initial caller cannot hear the Reverse Voice Over conversation. The private Intercom call continues until the Reverse Voice Over caller releases the key again. The initial call can be an outside call or an Intercom call.

Reverse Voice Over could help a salesman, for example, when placing a call to an important client. The salesman can talk with the client **and** give special instructions to a secretary - without interrupting the initial call.

When the keyset is idle, the Reverse Voice Over key functions the same as a Hotline key. A keyset's Reverse Voice Over key also shows at a glance the status of the associated extension:

When the key is. . .	The associated extension is. . .
Off	Idle
On	Busy or call ringing
Fast Flash	In Do Not Disturb

Note: When the keyset is idle, the Reverse Voice Over provides one button calling to the associated extension (like a Hotline key). An extension user cannot, however, use the Reverse Voice Over key to Transfer calls.

Ring Groups

Aspire S	Aspire M/L/XL
• Available - 8 Ring Groups.	• Available - 100 Ring Groups.
• Timer Class of Service requires software 5.91+	• Timer Class of Service requires software 5.91+

Description

Ring Groups determine how trunks ring extensions. Generally, trunks ring extension's only if Ring Group programming allows. For example, to make a trunk ring an extension:

- Assign the trunk and the extension to the same Ring Group
- In the extension's Ring Group programming, assign ringing for the trunk.

Any number of extensions and trunks can be in a specific group. The Aspire S system allows Ring Groups=1-8, In-Skin Voice Mail (102), or Centralized Voice Mail (103). The Aspire M/L/XL system allows Ring Groups=1-100, In-Skin Voice Mail (102), or Centralized Voice Mail (103).

If an extension has a line key for the trunk, Ring Group calls ring the line key. If the extension doesn't have a line key, the trunk rings the line appearance key. If an extension has a key for a trunk that is not in its ring group, the trunk follows Access Map programming.

Ring Tones, Selectable

Please refer to **Selectable Ring Tones** (page 185) for information on this feature.

Ringdown Extension, Internal/External

Aspire S	Aspire M/L/XL
• Available - 512 Hotline assignments and 50 extensions/virtual extensions with software 2.08-2.21; 66 extensions/virtual extensions with software 2.50 and higher.	• Available - 512 extensions/virtual extensions and 512 Hotline assignments.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.

Description

With a Ringdown Extension, a user can call another extension, outside number, or Abbreviated Dialing number by just lifting the handset. The call automatically goes through - there is no need for the user to dial digits or press additional keys. Ringdown Extensions are frequently used for lobby phones, where the caller just lifts the handset to get the information desk or off-site Reservation Desk.

After the Ringdown Extension user lifts the handset, ringdown occurs after a programmable interval. Depending on the setting of this interval, the extension user may be able to place other calls before the ringdown goes through.

The system allows each extension in the system to have a Ringdown Extension. All extensions can share the same dialing number, if desired.

Room Monitor

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Room Monitor lets an extension user listen to the sounds in a co-workers area. For example, the receptionist could listen for sounds in the warehouse when it's left unattended. To use Room Monitor, the initiating extension ***and*** the receiving extension must activate it.

When using keysets for monitoring, an extension user can only Monitor one extension at a time. However, many extensions can Monitor the same extension at the same time.

With single line phones, multiple SLTs can be programmed to be monitored by the same SLT. However, only one SLT can monitor another SLT at a time.

Room Monitor for Single Lines

This option enables you to monitor the room status through your single line telephone. Between keysets, the monitored room status is picked up by the phone's microphone and the activity is heard through the speaker of the monitoring keyset. Between single line phones, a user goes off hook on the monitored phone and, from another single line phone, dials a service code and the extension number. The activity of the area where the monitored phone is placed can then be heard at the monitoring phone. This service is available until the handset of the monitored telephone is placed on hook.

CAUTION

The use of monitoring, recording, or listening devices to eavesdrop, monitor, retrieve, or record telephone conversation or other sound activities, whether or not contemporaneous with transmission, may be illegal in certain circumstances under federal or state laws. Legal advice should be sought prior to implementing any practice that monitors or records any telephone conversation. Some federal and state laws require some form of notification to all parties to a telephone conversation, such as using a beep tone or other notification methods or requiring the consent of all parties to the telephone conversation, prior to monitoring or recording the telephone conversation. Some of these laws incorporate strict penalties.

Save Number Dialed

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Save Number Dialed permits an extension user to save their last outside number and easily redial it later on. For example, an extension user can recall a busy or unanswered number without manually dialing the digits. The system retains the saved number until the user stores a new one in its place.

Save Number Dialed saves in system memory a dialed number up to 24 digits. The number can be any combination of digits 0-9, # and *. The system remembers the digits regardless of whether the call was answered, unanswered or busy. The system normally uses the same trunk group as for the initial call. However, the extension user can preselect a specific trunk if desired.

Secretary Call (Buzzer)

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Secretary Call lets two co-workers alert each other without disturbing their work. To have Secretary Call, both co-workers must have keysets with Secretary Call buzzer keys. When a user presses their buzzer key, the system alerts the called extension by sending a splash tone and flashing the called extension's buzzer key. The called user can respond by placing an Intercom call to the calling party. The called extension's buzzer key continues to flash and the splash tone is heard until either user cancels the Secretary Call. A secretary could use this feature, for example, to get a message through to the boss in an important meeting. After being alerted, the boss could call the secretary when it's most convenient.

An extension can have Secretary Call keys for any number of extensions, limited only by the available number of programmable keys.

Secretary Call Pickup

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Secretary Call Pickup lets a keyset user easily reroute calls intended for a co-worker to themselves. By pressing a Secretary Call Pickup key, the user can have all calls to a co-worker's phone ring or voice-announce theirs instead. Secretary Call Pickup is a simplified type of Call Forward with Follow Me for employees that work closely together. This feature could be helpful to customer service representatives that must frequently cover each other's clients. When a representative leaves their desk, an associate could press the Secretary Call Pickup key to intercept all their calls.

An extension can have Secretary Call Pickup keys for any number of extensions, limited only by the available number of programmable keys.

Secure Set Relocation

Please refer to Maintenance (page 152) for information on this feature.

Selectable Display Messaging

Aspire S	Aspire M/L/XL
• Available - all telephones are able to use Selectable Display Messaging at one time.	• Available - all telephones are able to use Selectable Display Messaging at one time..

Description

An extension user can select a preprogrammed Selectable Display Message for their extension. Display keyset callers see the selected message when they call the user's extension. Selectable Display Messaging provides personalized messaging. For example, an extension user could select the message "GONE FOR THE DAY". Any display keyset user calling the extension may hear a DND signal and then see the message. See table below for a list of the standard messages.

An extension user can add digits for date, time or phone number after messages 1-8 and 10 (up to 24 characters). For example, an extension user could select the message "ON VACATION UNTIL" and then enter the date. Callers see the original message followed by the appended date. They would then be able to tell when the user was coming back from vacation. The system allows all phones to use the Selectable Display Messaging feature at the same time.

The default messages are:

No.	Message	Change "#" to...
1	IN MEETING UNTIL ##:##	Time (when meeting done)
2	MEETING ROOM - ########	Room Name or extension
3	COME BACK ##:##	Time (when returning)
4	PLEASE CALL ###########	11 digits (phone number)
5	BUSY CALL AFTER ##:##	Time (when returning)
6	OUT FOR LUNCH BACK ##:##	Time (when returning)
7	BUSINESS TRIP BACK ##/##	Date (when returning)
8	BUSINESS TRIP ##########	10 digits (where reached)
9	GONE FOR THE DAY	
10	ON VACATION UNTIL ##/##	Date (when returning)
11-20	MESSAGE 11-20	

Selectable Ring Tones

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

An extension user can change the way trunks or ICM calls ring their phone. Selectable Ring Tones allows an extension user to set up unique ringing for their calls. This is important in a crowded work area where several phones are close together. Because their phone has a characteristic ring, the user always can tell when it's their phone ringing.

Serial Call

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Serial Call is a method of transferring a call so it automatically returns to the transferring extension. Serial Calling saves transferring steps between users. For example, a Customer Service Representative (CSR) has a client on the phone who needs technical advice. The CSR wants to send the call to Tech Service, but needs to advise the client of certain costs when Tech Service is done. Rather than transferring the call back and forth, the CSR can use Serial Call to Technical Service and announce, "I have Ted on the phone. I need to talk to him again. Just hang up when you're done and I'll get him back."

Single Line Telephones, Analog 500/2500 Sets

Aspire S	Aspire M/L/XL
• Available - 16 single line telephones maximum.	• Available - **Aspire M/L:** 256 single line telephones maximum (may be restricted due to system power requirements). **Aspire XL:** 384 single line telephones maximum - requires software 4.0E+.
• Defining CODEC Filter settings available.	• Defining CODEC Filter settings available with software 1.04+.
• DTMF Dial Out Timer Added with software 4.0E+.	• DTMF Dial Out Timer Added with software 4.0E+.

Description

The system is compatible with 500 type (Dial Pulse) and 2500 type (DTMF) analog single line telephones (SLTs). You can install single line telephones as On-Premise or Off-Premise extensions. Single line telephone users can dial codes to access many of the features available to keyset users. With Single Line Telephones, you can have your system simulate PBX type operation.

When installing single line telephones, you must have:

- A port on an SLIU PCB for each single line telephone installed.
- (If you have 2500 sets) At least one block reserved on the NTCPU for analog extension DTMF reception.

CODEC Filter Data Setup Program Added

When **Program 82-07-01 : CODEC Filter Setup for Analog Station Ports** is set to "4 - Specified Data", the system will use the settings in **Program 82-09 : SLIU CODEC Filter Data Setup**.

These values should not be changed from their default settings unless directed by NEC'S Technical Service department.

The side tone of the SLIU is adjusted using all 16 values, however, special software is required in order to compute these values. The setting is not proportional to the gain level. To change these values, contact NEC's Technical Service department for assistance.

DTMF Dial Out Timer Added

A program is added for DTMF dialing, **Program 20-03-07 : System Options for Single Line Telephones**. When **Program 20-03-03 : System Options for Single Line Telephones - SLT DTMF Dial to Trunk Lines** is set to "0" (receive all digits before sending), the system will following the timers in Program 20-03-04 and 23-03-07.

The timer in **Program 20-03-04 : System Options for Single Line Telephones - Dial Sending Start Time for SLT or ARS** will reset when the user dials another digit.

The timer in **Program 23-03-07 : System Options for Single Line Telephones - Forced Dial Sending Start Time** will not reset when a digit is dialed. The user must finish dialing all the digits before this timer expires (Entries: 0-64800 seconds, Default: 0).

Single Line Telephones, Digital

Aspire S	Aspire M/L/XL
• Available - 24 single line telephones maximum.	• Available - **Aspire M/L:** 256 single line telephones maximum (may be restricted due to system power requirements). **Aspire XL:** 384 single line telephones maximum - requires software 4.0E+.

Description

The digital single line telephone (DSL) can be installed as On-Premise or Off-Premise extensions. Single line telephone users can dial codes to access many of the features available to keyset users. With single line telephones, you can have your system simulate PBX type operation. Each single line telephone installed must have a port on an ESIU PCB.

Soft Keys

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Each display telephone provides interactive soft keys for intuitive feature access. It is no longer necessary to remember feature codes to access the telephone's advanced features because the function of the soft keys change as the user processes calls. For example, just press a soft key to Page, Park a call, leave a message or Camp On to a busy co-worker.

Additional options allow you to "fine tune" the keyset's volume levels for handset receive and transmit, speaker volume, ringer and handset volume, and headset volume levels. You can also customize the point at which the built-in speakerphone switches from transmit to receive; a boon for noisy environments. The display telephones also have a contrast control for the LCD display.

Station Message Detail Recording

Aspire S	Aspire M/L/XL
• Available.	• Available.
• This feature requires a connection to the system using a CTA or CTU adapter, or through the serial, or LAN port on the Aspire ENTU. (The LAN port only provides information through LAN-capable programs, such as HyperTerminal. Printing of the SMDR information must be done from within that program.)	• This feature requires a connection to the system using a CTA or CTU adapter, or through the serial or USB port on the Aspire NTCPU (requires the USB driver which can be downloaded from the NEC Technical Support web site - ws1.necii.com). • The LAN port on the Aspire NTCPU can be used with 1.02+. (The LAN port only provides information through LAN-capable programs, such as HyperTerminal. Printing of the SMDR information must be done from within that program.)
• SMDR output includes Caller ID with software 4.0E+.	• SMDR output includes Caller ID with software 4.0E+.
• Enhanced SMDR displays require software 5.91+.	• Enhanced SMDR displays require software 5.91+.
• External Call Forward enhancement requires software 5.92+.	• External Call Forward enhancement requires software 5.92+.

Description

Station Message Detail Recording (SMDR) provides a record of the system's trunk calls. Typically, the record outputs to a customer-provided printer, terminal or SMDR data collection device. SMDR allows you to monitor the usage at each extension and trunk. This makes charge-back and traffic management easier.

SMDR provides the following options:

- **Abandoned Call Reporting**
  The SMDR report includes calls that rang into the system but were unanswered (i.e., abandoned). SMDR can include all abandoned calls or only those abandoned calls that rang longer than the specified duration. The Abandoned Call Report helps you keep track of lost business.

- **Blocked Call Reporting**
  When Toll Restriction blocks a call, you can have SMDR print the blocked call information. Or, you can have SMDR exclude these types of calls. With Blocked Call Reporting, you can better customize Toll Restriction for the site's application.

- **Customized Date Format**
  The SMDR header can show the report date in one of three formats: American, European or Japanese. Set the format for your preference.

- **Transferred Call Tracking**
  SMDR shows each extension's share of a transferred call. If an outside call is transferred among four extensions, SMDR shows how long each of the callers stayed on the call.

- **Data Call Tracking**
  Data Call Tracking can log the system's internal data calls. Since SMDR normally logs external (trunk) data calls, Data Call Tracking lets you get a complete picture of data terminal activity.

- **Digit Counting**
  With Digit Counting, SMDR can selectively keep track of toll calls. For example, if the digit count is nine, SMDR won't include toll calls within the home area code. Digit Counting permits SMDR to include only the types of calls you want to monitor.

- **Digit Masking**
  Digit Masking lets you "X" out portions of the number dialed on the SMDR report. A digit mask of seven, for example, masks out all exchange codes (NNXs) and local addresses. Digit Masking makes it easier to keep track of calling patterns, without having to interpret each individual number. You can also use Digit Masking to block out access and security codes.

- **Duration Monitoring**
  SMDR can include calls of any duration, or only those that last longer than the interval you specify. If you want to keep track of all trunk activity, use a short duration. To keep track of only significant usage, use a longer duration.

- **Extension Exclusion**
  You can selectively exclude extensions from the SMDR report. This ensures privacy for high-profile callers. For example, the company attorney negotiating a merger may not want his calls to show up on an in-house report.

- **PBX Call Reporting**
  If you system is behind a PBX, you can have SMDR monitor all traffic into the PBX or just calls placed over PBX trunks. The SMDR record can include all PBX calls (including calls to PBX extensions) or just calls that include the PBX trunk access code.

- **Serial and USB SMDR Communication**
  The system is compatible with both serial and USB SMDR devices. This gives you many SMDR output options. For example, you can output the SMDR report to a high speed printer or send it to disk through a PC's serial or USB port (requires the USB driver which can be downloaded from the NEC Technical Support web site - ws1.necii.com).

- **Trunk Exclusion**
  Use Trunk Exclusion to exclude certain trunks not subject to per-call charges (like WATS lines) from the SMDR report. This makes call accounting easier, since you review only those calls with variable costs.

- **Usage Summaries**
  SMDR can automatically print daily, weekly and monthly call activity summaries. Each summary includes the total number of regular trunk calls and ISDN trunk calls, and the costs for each type. The daily report prints every day at midnight. The weekly report prints every Sunday night at midnight. The monthly report prints at midnight on the last day of the month.

- **Extension Name or Number**
  The SMDR report can include an extension's name or extension number. Choose the method that makes it easier for you to track call usage.

SMDR Enhanced for Caller ID

Depending on your software version, the SMDR output is enhanced to include up to 16 or 24 characters of the Caller ID name information (depending on the view option selected in Program 35-02-18). You can select to display the Caller ID number or name or the DID number. If you wish to display the Caller Name in the “DIALLED NO./CLI” and “ACCOUNT" area, select "2" in the updated Program 35-02-15 and "1" in Program 35-02-17.

If the Caller ID name is not received, the area for Caller ID Name is left blank.

Enhanced SMDR Displays

The SMDR report has been enhanced to provide the following:

- The "DIALLED No./CLI" field can print up to 20 digits. A new program is added to determined whether the report shows the first 20 digits or the last 20 digits.
- The "Line" field on the SMDR can show both the trunk number (or trunk name) and the DID number based on the setting in Program 35-02-16.

External Call Forward Setting Display Enhanced

A program option (Program 35-02-20) for the SMDR reports is available which determines which information is displayed in the "STATION" area for a transferred call when the extension has Call Forward set with an Abbreviated Dial number as the destination. You can choose to display the extension number which **called** the extension with external Call Forward set or display the extension number which **has the external Call Forward set**.

This option only applies when Call Forward is set using a service code (Program 11-11-01~11-11-07) and the destination uses an Abbreviated Dial bin. It does not include Off-Premise or Centrex transfers.

Prior to this enhancement, the report would only display the extension number which called the extension with external Call Forward set.

Station Park

Please refer to **Park** (page 175) for information on this feature.

Station Relocation

Please refer to **Maintenance** (page 152) (Extension Data Swap) for information on this feature.

T1 Trunking (with ANI/DNIS Compatibility)

Aspire S	Aspire M/L/XL
• Not Available.	• Available • ANI/DNIS Compatibility is available. • This feature requires the Basic NTCPU (P/N 0891002) with the PAL upgrade or the Enhanced NTCPU (P/N 0891038) unless using software 4.0E or higher.

Description

The T1/PRI Interface PCB gives the system T1 trunking capability. This PCB uses a single universal slot and provides up to 24 trunk circuits. In additional to providing digital-quality trunking, the T1/PRI Interface PCB allows you to have maximum trunking capability with fewer PCBs. This in turn makes more universal slots available for other functions.

You can program each T1/PRI PCB for any combination of the following trunks:

- CO loop start
- CO ground start
- Direct Inward Dialing
- Tie lines [1]

The T1/PRI Interface PCB uses the first block of 24 consecutive trunks. For example, if you have an 8COIU PCB installed for trunks 1-8, the T1/PRI Interface PCB will automatically use trunks 9-32. If you have 8COIU PCBs installed for trunks 1-8 and 17-24, the T1/PRI PCB will use trunks 25-48. The T1/PRI Interface cannot use trunks 9-16 (even if available) since they are not part of a consecutive block of 24 trunks. Each T1/PRI PCB requires that 24 consecutive ports be available in the system even if not all the ports will be used otherwise the PCB will not function.

ANI/DNIS Compatibility

The system is compatible with telco's T1 Automatic Number Identification (ANI) and Dialed Number Information Service (DNIS) services. This is only for T1 - not for PRI at this time). A compliment to Caller ID service, ANI/DNIS Compatibility provides:

- **Selectable Receive Format**
  You can set up the system for compatibility with any combination of ANI, DNIS and Dialed Number (Address) data provided by the telco.
- **Flexible Routing**
  Based on the data received, the system can route the incoming ANI/DNIS call to:
  - An extension
  - An ACD or Voice Mail master extension number
  - The VRS and play a VRS message to the caller.
  - A Department Group pilot number
  - A trunk Ring Group

1. Two-wire (four-lead) type 1 tie lines (FIC TL11M) only.

- **Route According to DID Translation Table or Abbreviated Dial Bins**
  Calls can be routed based on either the number of digits defined in Program 22-09-01 (digits 1-8) or by digits entered in Abbreviated Dial bins in Program 13-04-01.
- **ANI/DNIS Data Displayed as Caller ID Data**
- **Data Error and Unanswered Call Handling**
  If a call can't be completed, send it to a predetermined Ring Group or play supervisory tones to the caller.

Tandem Ringing

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Tandem Ringing allows an extension user to have two telephones with one phone number. For example, extension 305 (the master phone) sets Tandem Ringing with extension 306. When extension 305 receives an incoming call, both extension 305 and 306 ring. Callers would dial the master extension number (extension 305 in this example). When either the master phone or slave phone is in use, the other phone cannot be used for outgoing calls - incoming calls, however, will ring the available phone.

The keyset must be paired with either an analog single line telephone or Aspire Wireless extension. It cannot be paired with another keyset, IP phone, or Cordless phone.

Tandem Trunking (Unsupervised Conference)

Aspire S	Aspire M/L/XL
• NTCPU provides 32 conference circuits, allowing each block to have any number conferences with any number of internal or external parties conferenced as long as the total number of conference channels used does not exceed the limit of 32.	• NTCPU provides 2 blocks of 32 conference circuits, allowing each block to have any number conferences with any number of internal or external parties conferenced as long as the total number of conference channels used does not exceed the block's limit of 32.
• Automatic conference on Hang Up available with software 2.10+.	• Automatic conference on Hang Up available with software 2.10+.
• Trunk Disconnect Continue and Disconnect Codes require software 4.0E+.	• Trunk Disconnect Continue and Disconnect Codes require software 4.0E+.
• Multiple Trunk Conference with 3 or more trunks requires software 4.93+.	• Multiple Trunk Conference with 3 or more trunks requires software 4.93+.

Description

Tandem Trunking allows an extension user to join two outside callers in a trunk-to-trunk Conference. (Depending on your software level, it may be possible to conference 3 or more trunks with Tandem Trunking.) The extension user can then drop out of the call, leaving the trunks in an Unsupervised Conference. The extension user that established the Conference is not part of the conversation. The Conference continues until either outside party hangs up. In addition, the extension user that set up the Conference can end the tandem call at any time.

The number of simultaneous Conference calls is limited by the number of Conference circuits in the system. Due to this fact, the maximum number of Conference calls cannot exceed the limits defined in the above table.

Tandem Trunking could help an office manager, for example, put two outside sales people in touch. The office manager could:

- Answer a call from one salesperson
- Place a call to the second salesperson
- Set up the trunk-to-trunk Conference
- Drop out of the call

The office manager could terminate the Conference at any time.

There are four methods for Tandem Trunking:

- **Method A - Set Up Without Transfer Key**
  An extension user can set up Tandem Trunking (Unsupervised Conference) by dialing a two-digit service code (#8) or a uniquely programmed Transfer key.
- **Method B - Tandem Trunking with Transfer Key**
  This method allows an extension user to easily set up an Unsupervised Conference with a call they have placed on Hold. It uses a uniquely programmed Transfer key to set up a tandem call.
- **Method C - Automatic Tandem Trunking on Hang Up**
  This method allows an extension user to easily set up an Unsupervised Conference without having to place the conference call on Hold. A Class of Service option is available, depending on your software version, which will allow or deny an extension user from automatically setting up a Conference/Tandem Trunking call upon hanging up the phone.
- **Method D - Automatic Tandem Trunking Setup to Abbreviated Dial Number**
  This method allows an extension user to easily set up an Unsupervised Conference with a call they have placed on Hold. A Class of Service option is available, depending on your software version, which will allow or deny an extension user from automatically setting up a Conference/Tandem Trunking call upon hanging up the phone.

Trunk Disconnect Continue/Disconnect Codes Added

The software enhances the forced trunk release option with the Tandem Trunking and DISA features. With older software, with **Program 24-02-07 : System Options for Transfer - Forced Release for Trunk-to-Trunk Transfer** or **Program 25-07-08 : System Timers for VRS/DISA - DISA Long Conversation Disconnect** enabled, users were not provided with an option to continue the call, if required.

With software 4.0E+, users can be provided with the option to use a Continue or Disconnect service code. The Continue service code will extend the conversation a programmed length of time. If the user enters the Disconnect service code, the call will be disconnected immediately.

Multiple Trunk Conference Allows 3 or More Trunks

The Tandem Trunking has been enhanced with the Multiple Trunk Conference feature. Previously, only two outside trunks could be connected for a trunk-to-trunk conference. This enhancement allows multiple trunks to be connected, the extension user can then drop out of the call, leaving the outside calls connected in an unsupervised conference.

This enhancement also allows the user which established the unsupervised conference to reenter the conversation. The user can then let the conversation continue or they can disconnect the trunks.

The number of simultaneous Conference calls is limited by the number of Conference circuits in the system. Due to this fact, the maximum number of Conference calls cannot exceed the limits defined in the above table.

There are two methods for Multiple Trunk Conferencing:

- **Method A - Set Up Without Transfer Key**
  An extension user can set up Multiple Trunk Conference (Unsupervised Conference) by dialing a two-digit service code (#8).
- **Method B - Multiple Trunk Conference with Transfer Key**
  This method allows an extension user to easily set up an Unsupervised Conference using a uniquely programmed Transfer key.
  Note: This operation is different than when only connecting 2 outside callers - the call can be on hold and, after calling the second party, the Transfer key can be pressed. With more than 2 outside calls, the CONF key must be pressed in order to connect the callers.

TAPI Compatibility

Aspire S	Aspire M/L/XL
• Available.	• Available with software 1.07 or higher.
• Third-party TAPI is available.	• Third-party TAPI requires the Basic NTCPU (P/N 0891002) with the PAL upgrade or the Enhanced NTCPU (P/N 0891038).

Description

The system has Telephony Applications Programming Interface (TAPI) capability. TAPI capability provides:

- Reduced TAPI Feature Set (see the Supported TAPI Commands chart below).
- Caller ID data to the PC for data base lookups and screen pops (see the Caller ID Data chart below).
- Telephone control (off-hook, on-hook and dialing).

The CTA or CTU adapter provides an interface that allows the user personalized control of the telephone system from a desktop or laptop PC when used in conjunction with a TAPI-compliant application. The telephone system and PC are connected by installing an adapter on the telephone keyset, allowing the PC user to access sophisticated communications services via the telephone lines.

In addition to a compatible system software version, you must also have:

- Aspire keyset telephone containing an RS-232-C CTA Adapter (P/N 0890058) / CTU Adapter (P/N 0890059) with TAPI compliant firmware.
- PC Driver for the CTA: PC running Windows 98 Second Edition or higher
- PC Driver for the CTU: PC running Windows 98 Second Edition, Windows 2000, or Windows XP
- A TAPI compatible Windows application

Tie Lines

Aspire S	Aspire M/L/XL
• Not Available.	• Available. • This feature requires the Basic NTCPU (P/N 0891002) with the PAL upgrade or the Enhanced NTCPU (P/N 0891038).

3

Description

Tie lines directly link a local telephone system with one or more remote systems. The link is independent of the telco's switched network. When a local system user seizes a tie line, they hear Intercom dial tone from the remote system. The user may then be able to:

- Dial extensions in the remote system
- Use the remote system's trunks for outgoing calls
- Access Common Abbreviated Dialing bins in the remote system
- Use the remote system's Internal and External Paging

The system provides connection for 2-wire (four lead, tip/ring) or 4-wire (eight lead, tip/ring/tip 1/ring 1) type tie line circuits. Using switches on the PCB, each circuit type can be set as Type I, II, III, IV, or V. Refer to the Aspire M/L/XL Hardware Manual for the PCB switch settings.

Tie Line Class of Service

Tie Line Class of Service provides features and dialing restrictions for incoming tie lines. This allows you to control the capabilities of callers dialing into your system. The tie line Class of Service options are:

- **First Digit Absorption**
  A tie line can ignore (absorb) the first digit received, which helps when setting up a tie line network. For example, your system can have tie lines to two other systems with the same extension numbering plan. Use the first digit to differentiate between the systems. Tie line callers can dial 3200-3456 for the first system's extensions and 4200-4456 for the second system's extensions. The receiving system ignores the first digit and routes calls correctly to the extension dialed (i.e., 4301 is received as 301).

- **Trunk Group Routing/ARS Access**
  When a tie line user calls the remote system, they may be able to dial 9 and place outside calls through the remote system. Any toll charges are incurred by the remote system. The call follows the remote system's Trunk Group Access or Automatic Route Selection - whichever is enabled

- **Trunk Group Access**
  Tie line callers may be able to access trunk groups in the remote system by dialing Programmable Function Key *02 and the trunk group number. This allows the callers to select a specific trunk group for an outgoing call. Trunk Group Access bypasses the remote system's Trunk Group Routing/ARS. As with dial 9 access, any toll charges are incurred by the remote system.

- **Common Abbreviated Dialing**
  The remote system's Common Abbreviated Dialing bins may be available to tie line callers. Use this capability to set up centralized Abbreviated Dialing control - or just save time when dialing.

- **Operator Calling**
  A tie line caller may be able to dial 0 for the remote system's operator.

- **Paging**
  Internal and External Paging may be available to tie line callers. This allows co-workers in adjacent facilities connected by tie lines, for example, to broadcast announcements to each other.

- **Direct Trunk Access**
  This option allows tie line callers to directly access a trunk for an outside call by dialing #9 and the trunk's number. Like Trunk Group Access, this bypasses the remote system's Trunk Group Routing/ARS. Any toll charges are incurred by the remote system.

- **Forced Trunk Disconnect**
  The Forced Trunk Disconnect option allows a tie line caller to disconnect (release) another extension's active outside call. The tie line caller can then place a call on the released trunk. Tie line callers should use Forced Trunk Disconnect only in an emergency, when no other trunks are available.

- **DISA/Tie Trunk Barge In**
  The DISA/Tie Trunk Barge In option allows a tie line caller to break into another extension's established call. This sets up a three-way conversation between the intruding party and the two parties on the initial call.

Tie Line Outgoing Call Restriction
You can selectively deny incoming tie lines access to your system's outgoing trunk groups. Incoming tie line callers could be able to access your outgoing WATS lines, for example, but not your DDD trunks. The system allows you to set up a restriction matrix for each of your incoming tie lines - for each of your outgoing trunk groups.

Tie Line Toll Restriction Class
Incoming tie lines can have a Toll Restriction Class and be subject to the system's toll restriction. For example, Toll Restriction can prevent users from dialing 1-900 calls. When an incoming tie line caller tries to use system trunks to dial a 1-900 service, Toll Restriction will deny the call.

Flexible Tie Line Service Compatibility
You can individually program tie lines for Dial Pulse (DP) or DTMF incoming or outgoing signaling. Outgoing tie lines can be either wink start or immediate start.

Wink Start Mode Does Not Require a DTMF Receiver
E&M trunks in wink start mode do not require a DTMF receiver for dial tone detection. It is detected with the wink. This frees the DTMF receivers for other use.

Time and Date

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Clock Adjustment program available with software 2.62+.	• Clock Adjustment program available with software 2.62+.

Description

The system uses Time and Date for:

- Central Office Calls (Access Maps)
- Class of Service (Class)
- Direct Inward Lines
- Display Telephones
- Fax Machine Compatibility
- Night Service (Automatic)
- Programmable Trunk Parameters
- Ring Groups
- Station Message Detail Recording
- System Reports
- Toll Restriction (Class)
- Trunk Group Routing
- Voice Mail
- Voice Response System

Using the Daylight Savings Setup program, you can determine whether the system should automatically adjust the system time for daylight savings time/standard time changes.

Clock Adjustment

The system can be programmed to automatically adjust the system clock on a nightly basis. This feature allows you to make adjustments should the system cabinet regularly lose or gain time.

3

Toll Restriction

Aspire S	Aspire M/L/XL
• Available- 15 Toll Restriction Classes and 26 extensions.	• Available-15 Toll Restriction Classes and 512 extensions.
• Additional programming options available to restrict outgoing calls.	• Additional programming options added to restrict outgoing calls with 1.03+.
• Additional Common Permit Table default entries available.	• Common Permit Table default entries changed with 1.03+.
• Enhanced tie line Toll Restriction requires software 2.65+.	• Enhanced tie line Toll Restriction requires software 2.65+.

Description

Toll Restriction limits the numbers an extension user may dial. By allowing extensions to place only certain types of calls, you can better control long distance costs. The system applies Toll Restriction according to an extension's Toll Restriction Class. The system allows for up to 15 Toll Restriction Classes.

Toll Restriction offers the following capabilities:

- **Common Permit Code Table**
  Use the Common Permit Code Table when you have numbers you want all Toll Restriction Classes to dial. To let all users dial 911, for example, put 911 in the Common Permit Code Table. The Common Permit Code Table overrides the Restrict Code and Common Restrict Code Tables. The system provides 10 tables, with 10 entries in each table. Each code is 4 digits max., using 0-9, #, * and FLASH (as a wild card).

- **Common Restrict Code Table**
  The Common Restrict Code Table lets you globally restrict certain numbers for all Toll Restriction Classes. To prevent all users from dialing directory assistance (411), for example, put 411 in the Common Restrict Code Table. Be sure you don't allow the codes you want to restrict in the Permit Code Table or the Common Permit Code Table. The system provides 10 tables, with 10 entries in each table. Each code is 4 digits max., using 0-9, #, * and FLASH (as a wild card).

- **Restrict Code Table**
  When you want Toll Restriction to allow most calls and restrict only selected calls, use the Restrict Code Table. To block only 1-900 calls, for example, enter 1900 in the Restrict Code Table. (If the same Toll Restriction Class has both Permit and Restrict Code Tables, the system restricts calls that you enter only in the Restrict Code Table. Calls entered in both tables are not restricted.) The system provides 4 tables, with 60 entries (restricted codes) in each table. A restricted code is 12 digits maximum, using 0-9, #, * and FLASH (as a wild card).

- **Permit Code Table**
  The Permit Code Table lets you set up Toll Restriction so that users can dial only selected (permitted) telephone numbers. Use this table when you want to restrict most calls. To allow all users to dial only area code 203, for example, enter 1203 in the Permit Code Table. 1 + 203 + NNX + nnnn are the only numbers users can dial. (If the same Toll Restriction Class has both Permit and Restrict Code Tables, the system restricts calls that you enter only in the Restrict Code Table. Calls entered in both tables are not restricted.) The system provides 4 tables, with 200 entries (permitted codes) in each table. A permitted code is 12 digits maximum, using 0-9, #, * and FLASH (as a wild card).
- **International Call Restriction**
  International Call Restriction lets you limit the international calls an extension user may dial. You can build a restrict table to prevent only certain calls, or you can build a permit table to allow only certain calls. To allow most international calls, use the *International Call Restrict Table*. To prevent most international calls, use the *International Call Allow Table*. The system provides 10 International Call Restrict tables with up to 4 digits in each table entry and 20 International Call Allow tables, with up to 6 digits in each table entry. Valid entries are 0-9, #,* and FLASH (for a wild card).

- **Toll Restriction for Abbreviated Dialing**
  Abbreviated Dialing can bypass or follow Toll Restriction. If you allow many users to program Abbreviated Dialing, consider Toll Restricting the numbers they dial. If only administrators can program Abbreviated Dialing, Toll Restriction may not be necessary. You can separately restrict Group and Common Abbreviated Dialing.

- **Call Digit Counting**
  Use Call Digit Counting to limit the number of digits local callers can dial. You can use this option to prevent users from accessing local dial-up services. For example, set the Maximum Number of Digits in Local Calls to 7 to limit local callers to dialing the exchange code (NNX) and local address (nnnn) only. The system provides 4 tables in which you can make entries for this option. The range is 4-30 digits.

- **Toll Call Digit Counting**
  With Toll Call Digit Counting, you can limit the number of digits long distance callers can dial. This lets you prevent callers from dialing extensively into long distance dial-up services. You can make four entries (4-30 digits).

- **Toll Free Trunks**
  Certain trunks can be completely unrestricted, such as the company president's Private Line. Users can place calls on Toll Free Trunks anytime -- to anywhere, without inadvertently being toll restricted.

- **PBX Call Restriction**
  Toll Restriction programming lets you enable/disable PBX Call Restriction and enter PBX access codes. You only need to do this if your system is behind a PBX and you have trunks programmed for behind PBX operation. Refer to PBX Compatibility feature for the specifics.

Additional Toll Restriction Programming

Prior to 1.03 software, it was possible for users to place a call unrestricted in the following instance:

1. An outside call rings the system, but the caller hangs up before the call is answered.
2. Telco detects the end of the call and returns to an idle status.
3. If a user seizes the trunk within a second or two after this status change, but before the telephone stops ringing, the user is able to place an unrestricted call. This is because the system still recognizes the trunk as an incoming call, but to the telco, this is a new outgoing call.
   The system only provides Toll Restriction on outgoing calls.

To prevent this from occurring, the system now provides additional programming options. This option is not available for ISDN, DID, or E&M trunks.

Additional Default Entries For Common Permit Code Table

Additional entries have been added to the default Common Permit Code Table. The default setting is now as follows:

- Table 1: 911
- Table 2: 1800
- Table 3: 1888
- Table 4: 1822 (new)
- Table 5: 1833 (new)
- Table 6: 1844 (new)
- Table 7: 1855 (new)
- Table 8: 1866 (new)
- Table 9: 1877 (new)

Tie Line Toll Restriction Enhanced

With older software, the system only provided the use of Program 34-08 to determine the toll restriction a tie line trunk would follow. This table only allowed 20 entries.

With software 2.65 or higher, a new program is available, **34-01-05 : E&M Tie Line Basic Setup - System Toll Restriction**. If this option is set to '0', the system will follow the setting in **21-05-13 : Toll Restriction Class - Restriction of Tie Line Calls** to determine whether or not the toll restriction setting in Program 34-08 is to be followed. If this option is set to '1', the system will follow the system toll restriction settings defined in Program 21-05-01 through 21-05-13.

Toll Restriction, Dial Block

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Toll Restriction Dial Block lets a user temporarily block an extension's Toll Restriction. This helps a user block his or her phone from being used by another person while they are away from their desk. A user would need to enter a 4-digit personal code to enable/disable this feature.

Dial Block can also be set by the system administrator. If Dial Block has already been set by an extension user, the supervisor can not release it. Additionally, if Dial Block has been set by the supervisor, and extension user can not release it.

Important: This function works by password and Class of Service control (the supervisor is not an assigned extension). If Dial Block is available for all Classes of Service, everyone may become a supervisor if they know the Dial Block password.

Toll Restriction Override

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

Toll Restriction Override lets a user temporarily bypass an extension's Toll Restriction. This helps a user that must place an important call that Toll Restriction normally prevents. For example, you could set up Toll Restriction to block 900 calls and then provide a Toll Restriction Override code to your attendant and executives. When the attendant or executive needs to place a 900 call, they just:

- Press CALL1, dial a service code and enter their override code.
- Press a line key or dial a trunk access code (e.g., 9 or #9 002).
- Place the 900 call without restriction.

You can assign a different Toll Restriction Override code to each extension. Or, extensions can share the same override code.

Toll Restriction Override will override ***all*** Toll Restriction programming. Walking Toll Restriction allows you to assign a Toll Restriction level for each user. When a call is placed using Walking Toll Restriction, the restriction for the call is based on the Toll Restriction level defined in Programs 21-05-xx and 21-06-xx.

Traffic Reports

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

The system provides the ability to send data to a PC connected to the Aspire. The telephone call traffic data for each extension is captured for use with the SMDR feature.

Call Traffic

The total of outgoing call frequency, outgoing call duration, call charge, incoming call frequency, answer frequency, incoming call duration, ringing duration for each line and extension, and abandon call frequency for each line is logged. The total of incoming calls, answer frequency, call duration for each line and extension, and abandon call frequency of each line is logged and the data is outputted to the PC. The system totals the hour, day, week, and month for each terminal and trunk number. This information is used by the SMDR feature. The extension which is totalled is determined by system programming. The system outputs this data to the PC for the total period.

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Transfer

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Transfer into Conference available.	• Transfer into Conference available with software 1.11+.
• Transfer to trunk ring group requires software 2.63+.	• Transfer to trunk ring group requires software 2.63+.
• Pressing Transfer key (15-07 or SC 851:06) places call on hold requires 4.0E+.	• Pressing Transfer key (15-07 or SC 851:06) places call on hold requires 4.0E+.
• Step Transfer for Auto Trunk-to-Trunk Transfer requires software 4.93+.	• Step Transfer for Auto Trunk-to-Trunk Transfer requires software 4.93+.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.

Description

Transfer permits an extension user to send (i.e., extend) an active Intercom or outside call to any other extension in the system. With Transfer, any extension user can quickly send a call to the desired co-worker. A call a user transfers automatically recalls if not picked up at the destination extension. This assures that users do not lose or inadvertently abandon their transfers. While a transferred call is ringing an extension the system can optionally play ringback tone or Music on Hold to the caller.

The system allows the following types of transfers:

- **Screened Transfer**
  The transferring user announces the call to the destination before hanging up
- **Unscreened Transfer**
  The transferring party extends the call without an announcement.
- **Extension (Department) Groups Transfer**
  The Transferring party sends the call to a Department instead of an extension.
- **Transfer Without Holding**
  A user presses a busy line key and waits for the call to complete. The system automatically sends them the call when the internal caller hangs up.

Automatic On-Hook Transfer Operation

With Automatic On-Hook Transfer, a Transfer goes through as soon as the transferring user hangs up. For example, extension 304 can answer a trunk, press HOLD, dial 305 and hang up. The system extends the call to extension 305. Without Automatic On-Hook Transfer, the call would stay on Hold at extension 304 when the user hangs up. To extend the call, the user at extension 304 would have to press CONF or a Transfer function key before hanging up.

Each method has advantages. Automatic On-Hook Transfer makes transferring calls easier. However, users have to be more aware of how they handle their calls on Hold. Without Automatic On-Hook Transfer, extending a call becomes a two-step operation - but separate from placing calls on Hold.

Prevent Recall of Transferred Call

The Class of Service program has an option that will allow you to prevent a Transferred call from recalling the originating extension if the call is not answered.

Transfer Call into Conference/Existing Call

This feature allows either a keyset or single line telephone user with Barge In capability the ability to transfer a call into an existing call. This call can be a 2-party call, a Conference call, or a Barge In Conference. The system allows Intercom, analog trunk, ISDN trunk and H.323 trunk calls to be transferred into a Conference call. This would allow, for example, an attendant to locate co-workers and then transfer them into an existing telephone meeting. There is no need for the attendant to locate all the parties at the same time and sequentially add them into the Conference.
This feature is not supported across a network or with S-Bus ports (this includes transferring an S-Bus call into a conference or transferring a call into a conference which includes an S-Bus port).

Transfer to Trunk Ring Group Available

Depending on your software version, it is possible to transfer a DID or trunk call to the trunk's defined ring group (defined in **Program 22-05-01 : Incoming Trunk Ring Group Assignment**). The trunk will then ring the defined extensions for the ring group.

This also allows the transferred call to ring over the External Paging (**Program 31-05 : Universal Night Answer/Ring Over Page**) so that an employee can answer the call from any available phone.

To enable this feature, the system has a program option, **Program 11-15-09 : Service Code Setup Administrative (for Special Access) - Transfer to Trunk Ring Group Code**. When a call is transferred using this service code, it's transferred to the ring group destination for that incoming trunk. For example, trunk 2 is in Ring Group 4. When the call is transferred using this service code, the trunk will ring all extensions programmed for Ring Group 4 or ring the External Paging Group for Ring Group 4, depending on how the system is programmed.

Program 22-04-01 : Extension Ring Group Assignment and **Program 22-05-01 : Incoming Trunk Ring Group Assignment** must be programmed to allow an extension access to the ring groups. If the call is not answered, it can overflow to the destination defined in **Program 22-08-01 : DIL/IRG No Answer Destination**.

This service code can also be used with the VRS. This provides the caller listening to the VRS message with the ability to transfer their call and have it ring the external page. The code the caller would dial is defined in **Program 25-06-02 : VRS/DISA One-Digit Code Attendant Setup**.

Transfer Key Can Place Call on Hold
Depending on your software version, while on a call, if the Transfer key (Pgm 15-07 or SC 851: 06) is pressed, the call will be placed on hold. With older software, pressing the Transfer key will not place an existing call on hold.

Step Transfer for Automatic Trunk-to-Trunk Transfer Feature Added
The Aspire enhances the Automatic Trunk-to-Trunk Transfer feature. If the destination to which a trunk has been transferred receives no answer (following the timer set in Program 24-02-12), the system can then automatically Step Transfer the call to a new destination. Up to 8 different destination numbers can be defined (Program 14-01-26).

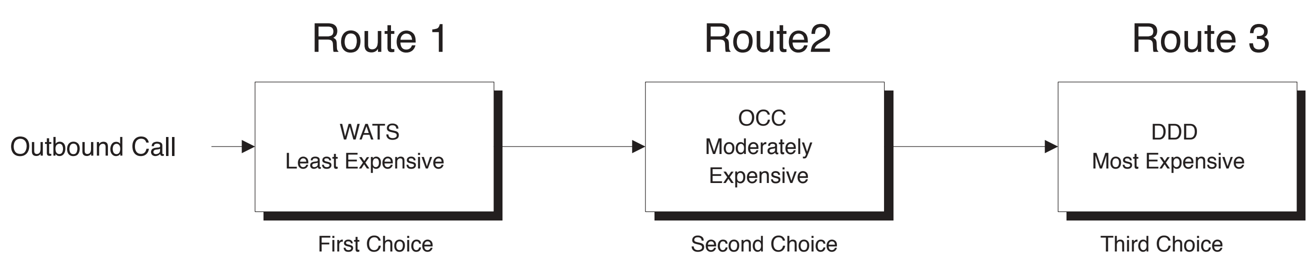
Trunk Group Routing

Aspire S	Aspire M/L/XL
• Available - 8 trunk groups and 8 routes.	• Available - 100 trunk groups and 100 routes.

3

Description

Trunk Group Routing sets outbound call routing options for users that dial the Trunk Group Routing code (9) for trunk calls. Trunk Group Routing routes calls in the order specified by system programming. If a user dials 9 and all trunks in the first group are busy, the system may route the call to another group. When you're setting up your system, Trunk Group Routing will help you minimize the expense of toll calls. For example, if your system has outbound WATS lines, OCC lines and DDD lines, use Trunk Group Routing to route calls to the WATS lines first.



Trunk Groups

Aspire S	Aspire M/L/XL
• Available - 8 trunk groups.	• Available - 100 trunk groups.

Description

Trunk Groups let you optimize trunk usage for incoming and outgoing calls. With Trunk Groups, users can have loop (rotary) keys for trunk calls. Incoming trunk group calls ring these loop keys. For outgoing calls, the user presses a loop key to access the first available trunk within the group. You set the access order in trunk group programming.

Loop keys give an extension user more available function keys, since the user doesn't need a separate line key for each trunk. The user only needs one loop key for each trunk group. This simplifies placing and answering calls.

Like Trunk Group Routing, Trunk Groups help you minimize the expense of toll calls. For example, if your system has outbound WATS lines, OCC lines and DDD lines, program the trunk group to route to the WATS lines first.

Priority	Type of Trunk
1	WATS
2	OCC
3	DDD

Trunk Queuing/Camp On

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.

Description

Trunk Queuing permits an extension user to queue (wait in line) on hook for a busy trunk or trunk group to become free. The system recalls the queued extension as soon as the trunk is available. The user does not have to manually retry the trunk later. Trunk Queuing lets the caller know when the call can go through. If the extension user does not answer the Trunk Queuing ring, the system cancels the queue request.

With Trunk Camp On, an extension user can queue (wait in line) *off hook* for a busy trunk or trunk group to become free. The caller connects to the trunk when the trunk becomes free. As with Trunk Queuing, the user does not have to manually retry the trunk later.

Any number of extensions may simultaneously queue or Camp On for the same trunk or trunk group. When a trunk becomes free, the system connects the extensions in the order that the requests were left.

Ultra CallAnalyst

Ultra CallAnalyst is an easy to use, graphically oriented software package that allows you to monitor and analyze phone calls, understand phone usage and cut costs. CallAnalyst tracks both incoming and outgoing calls accurately as well as the date and time of each call. If you need to track the incoming phone call with name and/or telephone numbers, CallAnalyst requires Caller ID services from the local phone company.

Ultra CallAnalyst increases productivity, facilitates billing and helps detect toll fraud and phone abuse. It also has powerful tabular (text) and graphical report generating capabilities. Reports include extension/line summaries, date/time and department summaries, longest/most expensive calls, most frequently called numbers, and other commonly used summaries. Ultra CallAnalyst also has the ability to automatically generate and Email reports to managers on set schedules. These reports can be used to analyze your telephone as a critical business communication tool, thereby, improving its effectiveness and helping you to reduce your telephone related costs.

CallAnalyst keeps track of:

- The date and time calls were made or received.
- The duration of each call.
- Which extension made or received the call.
- The CID/ANI, DNIS of the caller.
- The trunk or line numbers which handled the call.
- Account codes and authorization codes used for the call.

As an example; a report can be generated showing calling patterns by volume or duration on a color coded map of the United States. This can help a Customer Support, Sales Order, or Telemarketing business become more focused, more productive and cost effective.

CallAnalyst Enterprise Server

CallAnalyst Enterprise Server is a network version of Ultra CallAnalyst used for collecting and generating telephone usage reports for multiple telephone systems on a network. It is a scalable call accounting solution for the small to mid-size business.

CallAnalyst Enterprise Server is an easy to use, graphically oriented software package that allows you to monitor and analyze phone calls, understand telephone usage and cut costs. Ultra CallAnalyst tracks both incoming and outgoing calls accurately.

The application supports connection of heterogenerous (different type) telephone systems as well as the connected interfaces (serial or TCP/IP) and supports real time consolidation and reporting of call accounting data. If you need to track the incoming telephone traffic with calling name and/or telephone numbers, Ultra CallAnalyst requires Caller ID services from the local phone company.

Main Server Minimum PC Requirements
Hardware

- PC with Pentium 4 Processor
- 512 MB RAM
- SVGA Monitor with 1024 x 768 resolution
- 2 GB of free hard drive space
- CD-ROM drive (for software installation)
- Available serial port and RS-232 cable (if required)
- Network Interface Card (NIC)
- Printer (if required to print reports)

Software

- Windows NT 4.0 (workstation or server) w/SP6, 2000 Professional w/SP3, XP Professional w/ SP1 or 2003 Server
- MS SQL Server 2000 or MSDE (Microsoft Database Engine) for the database (MSDE is included on the application CD)
- Microsoft Internet Explorer 5.0 or higher (Internet Explorer is included on the application CD)

Network Client / Remote Site Reporting Client Minimum PC Requirements
Hardware

- PC with Pentium III Processor
- 256 MB RAM
- 1 GB of free hard drive space
- CD-ROM drive (for software installation)
- Network Interface Card (NIC)
- Available serial port and RS-232 cable (if required)

Software

- Windows NT 4.0 w/ SP6, or 2000 Professional w/ SP3, XP Professional w/ SP1 or 2003 Server
- Microsoft Internet Explorer 5.0 or higher (Internet Explorer is included on the application CD)

Universal Answer

Please refer to **Central Office Calls, Answering** (page 97) for information on this feature.

Voice Mail

Aspire S	Aspire M/L/XL
• Available - 72 NVM-Series voice mail ports available. External voice mail requires available analog station ports based on the number of voice mail ports connected, so this voice mail maximum is limited to 16 by the Aspire S. - Aspire Mail is not available. - 8 IntraMail ports (fixed ports 43-50) with software 2.50+.	• Available - 72 NVM-Series voice mail ports available. External voice mail requires available analog station ports based on the number of voice mail ports connected. - The Aspire Mail port requirements are based on the installed PCB (the 4VMSU requires 4 ports, 4VMDB requires 4 ports, 16VMSU requires 16 ports (and software 2.26+), 2FMSU requires 2 ports, 4FMSU requires 4 ports, 4FMDB requires 4 ports).
• The MSG key as Voice Mail key available.	• The MSG key as Voice Mail key feature is available with 1.11+ software.
• IntraMail options added with software 2.51+. - Ability to Select Voice Mail Port Selected for Message Notification/MW Lamps - External Transfer Available - Soft Key With Security Code Programming - Internal Message Notification Timer Lengthened	• IntraMail for the Aspire M only requires software 4.93+ and incorporates all prior IntraMail options from the Aspire S. IntraMail for the Aspire L/XL is not available.
• Intramail Directory Dialing and Multiple Greeting features requires software 2.64+.	
• Centralized voice mail with IntraMail available with software 4.94+.	• Centralized voice mail with Aspire Mail and IntraMail available with software 4.94+.
• Local in-skin voice mail (Aspire Mail, IntraMail) and centralized voice mail requires software 5.94+.	• Local in-skin voice mail (Aspire Mail, IntraMail) and centralized voice mail requires software 5.94+.

3

Description

The system is fully compatible with NEC's analog NVM-Series Voice Mail with Automated Attendant Systems. These systems provide telephone users with comprehensive Voice Mail and Automated Attendant features. Voice Mail ends the frustration and cost of missed calls, inaccurate written messages and telephone tag. This frees a company's busy receptionists and secretaries for more productive work.

Automated Attendant automatically answers the system's incoming calls. After listening to a customized message, an outside caller can dial a system extension or use Voice Mail.

Integrated Voice Mail enhances the telephone system with the following features:

- **Call Forwarding to Voice Mail**
  An extension user can forward their calls to Voice Mail. Once forwarded, calls to the extension connect to that extension's mailbox. The caller can leave a message in the mailbox instead of calling back later. Forwarding can occur for all calls immediately, for unanswered calls or only when the extension is busy. When a user transfers a call to an extension forwarded to Voice Mail, the call waits for the Delayed Call Forwarding time before routing to the called extension's mailbox. This gives the transferring party the option of retrieving the call instead of having it go directly to the mailbox.

- **Leaving a Message**
  Voice Mail lets a keyset extension user easily leave a message at an extension that is unanswered, busy or in Do Not Disturb. The caller just presses their Voice Mail key to leave a message in the called extension's mailbox. There is no need to call back later.

- **Transferring to Voice Mail**
  By using Transfer to Voice Mail, a keyset extension user can Transfer a call to the user's own or a co-worker's mailbox. After the Transfer goes through, the caller can leave a message in the mailbox.

- **Conversation Record**
  While on a call, an extension user can have Voice Mail record the conversation. The keyset user just presses the Voice Mail Record key; the ESL user dials a code. Once recorded, the Voice Messaging System stores the conversation as a new message in the user's mailbox. After calling their mailbox, a user can save, edit or delete the recorded conversation.

- **Personal Answering Machine Emulation**
  A keyset user can have their idle extension emulate a personal answering machine. This lets Voice Mail screen their calls, just like their answering machine at home. If activated, the extension's incoming calls route to the user's subscriber mailbox. Once the mailbox answers, the user hears the caller's incoming message. The keyset user can then:
  - Let the call go through to their mailbox
  - Intercept the call before it goes to their mailbox
  - Reject the call before it goes to their mailbox

- **Voice Mail Overflow**
  If Voice Mail automatically answers trunks, Voice Mail Overflow can reroute those trunks to other extensions when all Voice Mail ports do not answer or, with certain software, are busy. During periods of high traffic, this prevents the outside calls from ringing Voice Mail for an inordinate amount of time. There are two types of Voice Mail Overflow: Immediate and Delayed. With immediate overflow, calls immediately reroute to other extensions when all Voice Mail ports do not answer or, with certain software, are busy. With delayed overflow, calls reroute after a preset interval. Without any type of overflow, the outside calls ring Voice Mail until a port becomes available or the outside caller hangs up.

- **Message Center Mailbox**
  A Message Center Mailbox is a mailbox shared by more than one extension. Any keyset that has a Message Center Key for the shared mailbox can:
  - Listen to the messages stored in the shared mailbox.
  - Transfer calls to the shared mailbox.
  - Use many other Voice Mail features previously available only at an extension's individual mailbox.

  A Message Center Mailbox helps co-workers that work together closely - such as members of the same Department Hunt Group or ACD Group. For example, an ACD Group Supervisor can send important messages to the shared Message Center Mailbox, to which any ACD Group member can respond when time allows. Each ACD Agent's Message Center Key flashes when messages are waiting. (The Message Center Mailbox can be a mailbox for an installed, uninstalled or virtual extension.)

- **Voice Mail Caller ID**
  NVM-Series Voice Mail can use ANI/DNIS information to identify the outside caller that left a message in a user's mailbox. When the message recipient presses TI after hearing a message, they hear the time the message was sent and the outside telephone number of the message sender. Refer to **ANI/DNIS Compatibility** (page 191) for more information on setting up this feature.

Voice Mail Queuing
When accessing the voice mail, the system provides a voice mail queue. If all the voice mail ports are busy, any calls trying to get to the voice mail will be placed in queue. As the voice mail ports become available, the calls will be connected to the voice mail in the order in which they were received.

As the Voice Mail Queue follows Department Hunting programming, the queue can hold a maximum of 10 calls. If the queue is full or if the voice mail ports are not assigned to a Department Group, the calls will be handled as though there were no voice mail queuing feature enabled. The calls will either access voice mail if a port is available or they will receive a busy signal.

The Voice Mail Queuing feature does not work with the Conversation Record feature.

Park and Page
When an extension user is away from their phone, the voice mail provides a Park and Page feature which can let them know when they have a call waiting to be answered. To activate Park and Page, the subscriber records the Paging Message. To enable Park and Page, the user records a Paging message. Park and Page will then answer an incoming call and the system broadcasts the prerecorded Paging announcement. When the extension user hears the Page, they can go to any telephone and use Directed Call Pickup to intercept the call. Depending on how the subscriber wants Park and Page to operate, they can turn the Paging Message on or off. Refer to the Aspire Mail System Guide (P/N 17710SWGxx) for details on setting this feature.

MSG Key will Operate as Voice Mail Key
The system enhances a phone's MSG key function when connected to a system which has voice mail installed. When an extension receives a voice mail, the MSG key can be used to check the number of messages in voice mail, as well as call the voice mail to listen to the messages. If there is no Voice Mail Programmable Function Key defined (Program 15-07-01, code 77), the phone's Message Waiting LED will flash to indicate new messages.

This option is not available with a networked voice mail - the voice mail must be local.

In-Skin Voice Mail

The Aspire Mail/Aspire Mail Plus is an in-switch full-featured Voice Mail with Automated Attendant for the Aspire M/L/XL.

Aspire Mail / Aspire Mail Plus Specifications				
	0891032 Aspire Mail (Flash-based)	**0891037** Aspire Mail (Flash-based)	**0891033** Aspire Mail Plus (Hard disk)	**0891056** Aspire Mail Plus (Hard disk)
Ports:	2	4	4	8
Hours:	3	3	1400 (approximate)	1400 (approximate)
Mailboxes:	200	200	1000	1000
Messages (max):	7000	7000	7000	7000

The IntraMail is a plug-in "in-skin" full-featured, DSP-based integrated Voice Mail with Automated Attendant for Aspire S and Aspire M.

Aspire IntraMail Part Numbers and Capacities	
P/N 0892176	Aspire IntraMail 4 Port/4 Hour Kit Includes: ■ (1) P/N 0891003 DSPDBU (IntraMail PCB) ■ (1) P/N 0892175 4 Port/4 Hour CompactFlash Card with software. ■ (1) 0893820 Aspire IntraMail Literature Kit
P/N 0892180	Aspire IntraMail 4 Port/8 Hour Kit Includes: ■ (1) P/N 0891003 DSPDBU (IntraMail PCB) ■ (1) P/N 0892179 4 Port/8 Hour CompactFlash Card with software. ■ (1) 0893820 Aspire IntraMail Literature Kit
P/N 0892178	Aspire IntraMail 8 Port/8 Hour Kit Includes: ■ (1) P/N 0891003 DSPDBU (IntraMail PCB) ■ (1) P/N 0892177 8 Port/8 Hour CompactFlash Card with software. ■ (1) 0893820 Aspire IntraMail Literature Kit
P/N 0892182	Aspire IntraMail 8 Port/16 Hour Kit Includes: ■ (1) P/N 0891003 DSPDBU (IntraMail PCB) ■ (1) P/N 0892181 8 Port/16 Hour CompactFlash Card with software. ■ (1) 0893820 Aspire IntraMail Literature Kit
Mailboxes:	Station Mailboxes = 128 Routing Mailboxes = 16 Master Mailboxes = 16 (Only 8 of the 16 Master mailboxes are accessible in Aspire S.) Total Mailboxes = 160
Programming Interface:	Aspire S telephone programming or Aspire PCPro software 2.30 or higher.
Remote Programming:	Access Via HTML-based Aspire WebPro or using customer-provided modems with Aspire PCPro.
Voice Storage Media:	Flash Card (on IntraMail PCB)
Languages:	1 (English Mnemonic)

These voice mails can answer incoming calls and route them quickly and efficiently. Integrated Voice Mail features include Conversation Record, Answering Machine Emulation, and Caller ID with Return Call. Interactive Soft Keys guide the display telephone user through the extensive voice mail feature set.

IntraMail for Aspire S

The Aspire S IntraMail port assignments are fixed and should not be changed. The following is the default IntraMail:

- **Program 11-07-01 : Department Group Pilot Numbers**
  Master number is 700.
- **Program 16-02-01 : Department Group Assignments for Extensions**
  IntraMail ports assigned to Department Group 8.
  343=Order 43
  344=Order 44
  345=Order 45
  346=Order 46
  347=Order 47
  348=Order 48
  349=Order 49
  350=Order 50
- **Program 15-02-26 : Multi-Line Telephone Basic Data Setup, MSG Key Operation Mode**
  Extensions 301-308 are defined as voice mail keys (1).
- **Program 15-03-01 : Single Line Telephone Basic Data Setup - SLT Signaling Type**
  Set for DP (0).
- **Program 15-03-03 : Single Line Telephone Basic Data Setup - Terminal Type**
  Set to Special (1).
- **Program 45-01-01 : Voice Mail Integration Options - Voice Mail Department Group Number**
  IntraMail Department Group is Group 8.

IntraMail for Aspire M

The Aspire software now provides support for the IntraMail. Previously, this was only available for the Aspire S system. With this change, the 47-xx programs used for IntraMail are now available with the Aspire M. Refer to the IntraMail System Guide (P/N 0893240) for complete details on setting up the IntraMail.

The boot code for the Aspire system must be at the current level of 1.20 for IntraMail to function. Refer to the Boot Code Update Instructions, P/N 0893211 for details on updating. The current version number can be displayed with history enabled (DIMU) after rebooting the system. The version number will show on the first line of the display. Contact NEC’s Sales Support in order to obtain an boot update card if needed.

If the IntraMail will not be used, make sure **Program 47-01-17 : IntraMail System Options - Start of IntraMail Port** is set to "0" (this is the default setting). If there is an entry in this program, 8 ports will be reserved for IntraMail usage. This could prevent the system from reaching the full port capacity. This could be important especially when using the 64-port CPU (P/N 0891002).

IntraMail System Requirements

- Aspire M System Software version 4.93 or higher.
- 64-Port NTCPU (P/N 0891002) with basic factory-installed PAL chip or with the Feature Upgrade PAL chip (P/N 0891039)
  IntraMail will not work with the Enhanced NTCPU (P/N 0891038).
- 8 Unassigned Ports in the Aspire System
  Using IntraMail reduces the number of available ports in the system.
- IntraMail Kit

Unlike the Aspire S, the IntraMail ports in the Aspire M must be defined in system programming.

Notes:

- The VRS cannot be used when the IntraMail CompactFlash card is installed.
- An external voice mail and the IntraMail cannot work simultaneously.
- The Aspire Mail (VMSU/FMSU PCBs) and IntraMail cannot work simultaneously.
- The IntraMail will not work automatically as with the Aspire S. Programs must be defined for port usage.
- When the DSPDB is used for IntraMail, the daughter board does **not** provide any additional DSP resources (for DTMF Receivers, Caller ID Receivers, or Call Progress Tone Detection).
- The IntraMail does not work with the Enhanced NTCPU (PAL-B). If switching from an Enhanced NTCPU PAL chip to a Standard or Feature Upgrade for the IntraMail feature, ***the system must be cold started due to the port number differences*** or the IntraMail will not function correctly.
- PVMU stops when IntraMail is used. (The PVMU is not available in the U.S.)

IntraMail: Ability to Select Voice Mail Port Selected for Message Notification/MW Lamps

The Aspire S can select an available voice mail port which can be used the Message Notification or MW Lamp update routines.

IntraMail: External Transfer Available

The software provides the ability for the IntraMail to perform an external transfer. This allows the IntraMail to route an incoming Automated Attendant call out of the Aspire system on a new trunk based on an Abbreviated Dial number stored in a Dial Action Table.

IntraMail: Soft Key With Security Code Programming

The software provides a soft key when programming the security code. This soft key allows a user to select OK/CANCEL following an entry of a new code.

IntraMail: Internal Message Notification Timer Lengthened

When Message Notification places a call out, the system will wait up to 30 seconds for ringback, reorder, or busy tone from the trunk. If detected, notification call out processing begins normally. If not detected, the system abandons the call and decrements the Ring No Answer (RNA) count. In older software versions, the system would wait 15 seconds. This could cause notification callbacks to be inadvertently abandoned.

IntraMail: Directory Dialing

This feature requires the IntraMail Utility 1.2. The utility and the instruction sheet for updating the utility are available for downloading from the NEC Technical Support Site (ws1.necii.com). This site requires user registration (contact NEC Sales Support for details).

Directory Dialing allows an Automated Attendant caller to reach an extension by dialing the first few letters in the extension user's name. With Directory Dialing, the caller does not have to remember the extension number of the person they wish to reach — just their name. Here's how Directory Dialing works:

1. When the Automated Attendant answers, it sends the call to a Directory Dialing Mailbox. (Optionally, the caller may be asked to dial a digit to access Directory Dialing.)

2. The Directory Dialing Mailbox plays the Directory Dialing Message which asks the caller to dial letters for the name of the person they wish to reach.
3. The caller dials the letters for the person's name plus #. They can dial by first name or last name, depending on how the Directory Dialing Message was recorded and the Directory Dialing Mailbox was set up.
4. IntraMail searches the list of programmed extension names for a match of the caller-entered letters.
5. Voice prompts announce the first three matches, and allow the caller to dial a digit (1-3) to reach one of the announced matches. Additionally, the caller can dial 4 to hear additional matches (if any).
6. The caller dials the digit for the extension they wish to reach, and IntraMail sends the call to that extension. The call is sent as a Screened or Unscreened transfer, depending on programming.

For callers to use Directory Dialing, the system must have a name programmed for each extension (up to 15 characters, A-Z, using upper and lower case letters). Each extension should also have a name recorded in their Subscriber Mailbox. In addition, each extension used by Directory Dialing must be installed and must have their Subscriber Mailbox active (Personal or Group).

An outside caller can route to a Master Mailbox or Routing Mailbox programmed as a Directory Dialing Mailbox from:

- The Answer Table's Answer Schedule Override mailbox, Default mailbox, or Routing mailbox.
- A GOTO action in the Dial Action Table of a Call Routing Mailbox.

3

IntraMail: Multiple Greetings

This feature requires the IntraMail Utility 1.2. The utility and the instruction sheet for updating the utility are available for downloading from the NEC Technical Support Site (ws1.necii.com). This site requires user registration (contact NEC Sales for details).

The mailbox subscriber can record up to three separate greetings and make any one of the three active. When a caller leaves a message in the subscriber's mailbox, they hear the active greeting. This allows the subscriber, for example, to record separate greetings for work hours, after work, and during vacation. Instead of rerecording their greeting when they leave the office, they can just activate the "after work" greeting instead.

If the active greeting has not been recorded, a caller leaving a message in the subscriber mailbox will hear, "*At the tone, you can leave your message for (extension number or name).*"

Multiple Greetings requires IntraMail Voice Prompts version 1.2 or higher.

Centralized Voice Mail Can Use Aspire Mail and IntraMail

With software 4.94 or higher, the Aspire Mail and IntraMail cards can be used with Centralized Voice Mail. Previously, only external voice mails could be used with this feature.

For Centralized Voice Mail, the voice mail software must be version 11.08.05 or higher and all systems connected to the network must be updated to system software 4.94 or higher.

When the Aspire Mail is used for Centralized Voice Mail (CVM), the phones on the networked system can also have their displays updated using the NSL protocol. This allows the users to press the Soft Keys associated with the display to proceed through the voice mail menus. IntraMail does not support this operation and will function like an external voice mail on the networked system.

In order to send NSL protocol over the network, **Program 45-01-10 : Voice Mail Integration Options - NSL Protocol Support** must be enabled in order for the display to use the NSL protocol.

Prior to software 5.94, **Program 45-01-01 : Voice Mail Integration Options - Voice Mail Department Group Number** and **Program 45-01-08 : Voice Mail Integration Options - Networked Voice Mail Department Group Number** must both be programmed with the same group number. Otherwise, the audio path may not connected properly.

Using software 5.94+, a networked system can use a local in-skin voice mail (Aspire Mail, IntraMail) when centralized voice mail is set up (with previous software, the centralized voice mail ports would stop providing ring voltage and the call initiation protocols).

With this type of setup, **Program 45-01-01 : Voice Mail Integration Options - Voice Mail Department Group Number** has to be defined with the local voice mail's group number. This entry will be used to access the voice mail when the MSG key is pressed and in any other instance where the local voice mail would be used. When you wish to use centralized voice mail as well (as defined in **Program 45-01-08 : Voice Mail Integration Options - Networked Voice Mail Department Group Number**, then the user would need to dial the master number for the centralized voice mail.

The system will provide lamps and protocol for both the local and centralized voice mails (however, there is no indication with the lamping as to which voice mail has the new message).

Notes:

- When using the Aspire Mail for Centralized Voice Mail with an IP network, DTMF signaling is required. Set **Program 84-06-10 : VOIPU Setup : DTMF Behavior** to "1" (In-Band). An entry of "2" (Out-of-Band) can also be used when the signaling is supported by the VoIP protocol. Set **Program 84-12-31 : H.323 Phone CODEC Information Basic Setup - DTMF Relay Mode** to "1" (RFC2833).
- When the Aspire Mail is used for Centralized Voice Mail, the transfer protocol should be changed due to processing speed differences between the system and the Aspire Mail. In the Aspire Mail programming for the Internal Transfer screen (**Main Menu - Customize Database - System Options**), the Transfer String should be changed from the default entry of "F" to "FS".
- Unscreened transfers from the voice mail over the network with Centralized Voice Mail do not recall to the voice mail if not answered. This applies to internal or external voice mails.

Refer to the Aspire Networking Manual (P/N 0893207) for details on programming the Networking feature.

Refer to the Aspire IntraMail System Guide-P/N 0893240 or the Aspire Mail System Guide-P/N 17710SWG05 for complete details on programming the voice mail.

Voice Over

Aspire S	Aspire M/L/XL
• Available.	• Available.

Description

- Important -
Do not use Voice Over to a user on speakerphone as the conversation may be heard by the outside party.

Voice Over lets a user interrupt a keyset extension user busy on another call. With Voice Over, the busy keyset extension user hears an alert tone followed by the voice of the interrupting party. The keyset extension user can respond to the interrupting party without being heard by the original caller. If desired, the keyset extension user can easily switch between their original caller and the interrupting co-worker. The original caller and the interrupting party can never hear each other's conversation.

Voice Over could help a lawyer, for example, waiting for an urgent call. While on a call with another client, the lawyer's paralegal could announce the urgent call as soon as it comes in. The lawyer could then give the paralegal instructions how to handle the situation - all without the original client hearing the conversation.

Either a keyset or 500/2500 set user can initiate a Voice Over, but only a keyset user can receive a Voice Over.

To enable Voice Over, a keyset should have a function key programmed for Voice Over. In addition to one- touch Voice Over operation, the key shows the Voice Over status as follows:

When the key is . . .	You are . . .
Off	Not using Voice Over
Flashing	Listening to the interrupting party
On	Responding to the interrupting party

Voice Response System (VRS)

Aspire S	Aspire M/L/XL
• Available - 8 Channels.	• Available - 16 Channels.
	• VRS available with basic NTCPU requires software 2.00+.
• Using VRS with custom music files for MOH requires 5.15+.	• Using VRS with custom music files for MOH requires 5.15+.
• VRS Call Attendant provided with software 5.92+.	• VRS Call Attendant provided with software 5.92+.

Description

The DSP daughter board provides the option for Voice Response System (VRS) which gives the system voice recording and playback capability. The VRS CompactFlash card provides up to 48 system messages (General Message, Automated Attendant greetings, ACD messages, and the 900 Preamble). In addition, the Personal Greeting and Park & Page options can have up to 200 messages (note that the Park & Page feature uses 2 messages). This enhances the system with:

- **General Message** - provides a prerecorded message to which any user can listen
- **Personal Greeting** - lets an extension user record a message and forward their calls. Callers to the extension hear the recorded message and are then redirected.
- **Park and Page** - parks a call at an extension and automatically pages the user to pick it up
- **Automated Attendant (Operator Assistance)** - answers incoming calls, plays a greeting to the caller and then lets the caller directly dial a system extension
- **ACD Messages** - provides announcement and overflow messages for ACD groups
- **Transfer to the VRS -** any extension user can Transfer their outside call to the VRS
- **Voice Prompting Messages** - plays call and feature status messages to users
- **900 Preamble** - alerts callers using 900 lines of the cost and features of the “pay-per-call” service
- **Time, Date and Station Number Check** - lets a keyset extension user quickly hear a recording for the time, date, or the extension’s number.

VRS Messages

The VRS allows you to record up to 48 VRS messages. You allocate these messages for Automated Attendant greetings, the General Message, ACD messages and the 900 Preamble message. The total storage time for all VAU messages is approximately 45 minutes. The maximum duration for any type of message is 2 minutes - this is not programmable. VRS messages are battery backed up.

Any on-premise extension caller can listen, record and erase VRS Messages (unless restricted in programming). DISA and DID callers can listen and record VRS messages (unless restricted in programming).

General Message

A General Message is a prerecorded message available to all callers. A General Message typically contains important company information that all employees should hear. To hear the General Message, an employee can go to any keyset and press 4 (for General Message). You can restrict the ability to record the General Message in an extension's Class of Service. This allows you to give recording capability to the System Administrator or Communications Manager, for example, but not any employee. The MW LED at each telephone flashes when a new General Message is recorded. Once the extension user listens to the message, the MW LED goes out.

Personal Greeting

Personal Greeting allows an extension user to record a message and forward their calls. Callers to the extension hear the recorded message and are then forwarded to the new destination. The Personal Greeting and Park & Page options can have up to 200 messages total (note that the Park & Page feature uses 2 messages). With Personal Greeting, an extension user can add a personal touch to their Call Forwards. For example, a user can record:
"Hi. This is John Smith. I'll be out of the office today. In my absence, Mary Jones can answer all your questions. Please hold on for Mary."

After they record their Personal Greeting, the extension user chooses the condition that will activate Personal Greeting. Personal Greeting will activate for:

- Calls to the extension when it is busy or not answered
- All calls immediately
- Calls to the extension that are unanswered

The extension user then selects the destination for their calls. The choices are:

- A co-worker's extension
- Personal Greeting only (without forwarding)
- The extension user's own subscriber mailbox (if Voice Mail (DSP) is installed)
- Off-Premise via Common Abbreviated Dialing

In addition, the user can have Personal Greeting activate automatically for all calls, just CO (trunk) calls or just Intercom calls. When the user implements Personal Greeting for all calls, the system plays the greeting and reroutes:

- Calls transferred from the Automated Attendant (OPA)
- DISA calls ringing the extension
- DID calls ringing the extension
- Direct Inward Lines (DILs) ringing the extension
- Intercom calls

With Personal Greeting for only CO (trunk) calls, the system reroutes all of the calls listed above except Intercom calls.

Unique Personal Greeting Conditions
If a call comes into the extension when there are no VRS ports available to play the Personal Greeting, the system forwards the call without playing the recorded message to the caller.
If an extension has Personal Greeting (RNA) enabled, Intercom calls that voice announce are not subject to Personal Greeting rerouting.
Personal Greeting does not reroute normal Ring Group calls. Calls transferred from a co-worker or Voice Mail route to the forwarding destination without listening to the Personal Greeting.

Park and Page

When an extension user is away from their phone, Park and Page can let them know when they have a call waiting to be answered. The Personal Greeting and Park & Page options can have up to 200 messages total (note that the Park & Page feature uses 2 messages). To enable Park and Page, the user records a Personal Greeting along with an additional Paging announcement. Park and Page will then answer an incoming call and play the Personal Greeting to the caller. The caller then listens to Music on Hold (if available) while the system broadcasts the prerecorded Paging announcement. When the extension user hears the Page, they can go to any telephone and use Directed Call Pickup to intercept the call.

For example, John Smith could record a Personal Greeting that says:
"Hello, this is John Smith. I am away from my phone right now but please hold on while I am automatically paged."

The prerecorded Paging announcement could say:
"John Smith, you have a call waiting on your line."

The incoming caller hears the first message and listens to Music on Hold while the system broadcasts the second message. John Smith could then walk to any phone and pick up his call. If John doesn't pick up the call, the Page periodically repeats.

Park and Page follows the rules for Personal Greeting for All Calls, immediately rerouted. This means that Park and Page will activate for ringing Intercom calls, DID calls and DISA calls. It will also activate for calls transferred from the Automated Attendant. Additionally, calls from the Automated Attendant follow Automatic Overflow routing if not picked up. Park and Page will activate for transferred outside calls but not play the Personal Greeting to the caller. If a call comes in when the specified Page zone is busy, the system broadcasts the announcement when the zone becomes free.

Automated Attendant (Operator Assistance)

Automated Attendant automatically answers outside calls, plays a prerecorded greeting and then lets the outside callers directly dial system extensions, Department Calling Groups and Voice Mail. Automated Attendant provides immediate answering and routing of outside calls without the need for an operator or dispatcher. Automated Attendant provides:

- **Single Digit Dialing**
  Single Digit Dialing allows Automated Attendant callers to dial extensions, Department Calling Groups, and Voice Mail by pressing a single digit. For example, your Automated Attendant can greet calls with, *"Thank you for calling. To place an order, dial 1. To check on an existing order, dial 2. To speak with an operator, dial 0."* You can set up single digit dialing for each VRS Message programmed to answer outside calls via the Automated Attendant. This allows you to set up day/night/holiday greetings or unique greetings for each incoming trunk. (Keep in mind that, with a default system, if you assign destinations to digits 3, 4 and 5, outside callers will not be able to dial system extensions.)

- **Simultaneous Call Answering**
  With VRS installed, the Automated Attendant can answer up to 16 calls simultaneously.

- **Flexible Routing**
  The outside caller can directly dial any system extension, Department Calling Group or Voice Mail. If the caller dials a busy extension, Automated Attendant allows them to dial another extension or wait for the busy extension to become free.

- **Automatic Overflow**
  Automatic Overflow can automatically redirect a call if it can't go through. This can happen if all VRS ports are busy, if the called extension doesn't answer, or if the caller misdials or waits too long to dial. (This would occur if the caller is using a dial pulse telephone.) When the call overflows, it rings a designated Ring Group or the Voice Mail system.

- **Programmable Automated Attendant Greetings**
  You can record a different greeting for each trunk answered by the Automated Attendant. The greetings can be different in the day, at night or on holidays or weekends. You can also have a special greeting if the caller misdials. You record the greetings just the way you want. For example, "*Dial the three-digit extension number you wish to reach, dial 500 for Sales or dial 600 for Customer Service*." When assigning and recording Automated Attendant greetings, you can choose among the 48 VRS messages.

VRS Waiting Message

Using VRS Waiting Message, the system can automatically answer an incoming trunk call first (either a normal trunk or one designated for a department group) to let the outside caller hear a recorded message when the call is not answered in a programmed period of time. With this feature, the call keeps ringing at the same destination until it is answered or until other programming, takes affect.

This feature can use up to two messages for an incoming call and the duration between the messages is programmable. These messages will be repeated and, between these messages, either ring back tone or Music on Hold can be played.

This feature has two different modes:

- **Permanent Mode**
  This mode sets the feature using system programming and is available for the following types of calls.

  A. Normal Incoming Call
  When the call is not answered or a user presses the VRS Waiting Message function key, this feature will be initiated. The waiting message will be played until other no-answer program (transfer to another incoming ring group, disconnect, etc.) takes affect.

  B. Designated Call for the Department Group
  When a department group receives a call from a DID, DIL, DISA or E&M trunk and all terminals in the group are busy, the call will be put in a queue and VRS Waiting Message will be also be initiated. The waiting message will be played until other no-answer program (transfer to another incoming ring group, disconnect, etc.) takes affect or a terminal becomes available to receive the department call.

- **Manual Mode**
  This mode can be programmed by pressing the "VRS Waiting Message" function key from a KST to set this feature for each incoming ring group. This mode can be used for normal incoming calls only.

Transfer to the VRS

Any extension user can Transfer their outside call to the VRS. This lets their caller take advantage of the Automated Attendant's extensive routing capabilities. To Transfer the call, the user simply places the call on Hold, dials the unique VRS service code (set up in system programming) and hangs up.

Voice Prompting Messages

The VRS feature provides the system with Voice Prompting Messages. These Voice Prompting Messages tell the extension user the status or progress of their call. For example, if a user calls extension 300 when it is busy, they hear, "*Station 300 is busy. For Callback, dial 2.*"

900 Preamble

If the system has trunks that are part of a 900 (caller paid) service, the VRS can automatically play a prerecorded message when a user answers the call. This prerecorded message should describe the 900 service features and cost. The 900 Preamble ensures that the caller is always aware that they have accessed a 900 "pay- per-call" service. A system user cannot converse with the caller until the preamble message ends. If the caller hangs up before the message completes, they are not charged for the call. If the caller waits for the message to end, they can talk to a system user and call charging begins. The system will answer as many 900 calls as there are available VRS ports. If a 900 calls comes in when all VRS ports are busy, the call will not appear on an extension until a VRS port is available.

You can also use the 900 Preamble message to set up an *Auto-Answer with Greeting* application. When a receptionist answers a call, the VRS can play a preamble message such as, "Welcome to ABC Company. How can I help you?" When the caller replies, the receptionist answers, "One moment please," and quickly extends the call to the desired party. This ensures that all incoming calls are answered quickly, courteously and consistently.

Time, Date and Station Number Check

If the system has a DSP daughter board installed for VRS, any keyset user can find out the time, date or the extension's number while their phone is idle (on hook). The time and date check saves the user time since they don't have to look for a clock or calendar. Hearing the extension number conveniently identifies non-display keysets. To find out their extension number, the user presses 6 (for **N**umber). To listen to the time and date, the user presses 8 (for **T**ime).

Available with 64-Port Basic NTCPU

Prior to 2.00 software, the VRS feature required the NTCPU to be either:

- The Feature Upgrade PAL chip (P/N 0891039) with the 64-Port Basic NTCPU (P/N 0891002) OR
- The Enhanced NTCPU (P/N 0891038)

With software 2.00 or higher, and an update of the system's boot code, the VRS feature is available with the 64-port Basic NTCPU (no Feature Upgrade PAL chip required).

Note that the VRS feature requires a DSPDB be attached to the NTCPU with the optional VRS flash card installed. Although the DSPDB is recognized for this feature, it ***will not*** provide any additional tone resources (DTMF receivers, Caller ID receivers, or call progress tone detection).

Refer to the Boot Code Update Instructions, P/N 0893211, for details on installing the updated boot code.

VRS for Music on Hold

A user can now save their own music to the DSPDBU VRS to use as the Music on Hold source.

When using the VRS for Music on Hold, the music file must be CCITT μ-law 8kHz 8-Bit Monaural PCM data. This type of file can be easily encoded using the Windows Sound Recorder application or similar software). Any other type of formatted music file will be played as silence.

The music file must be saved in the **\VM\OGM\3\1\9** directory of the VRS compact flash card and it must be named "**Gxx.WAV**". The "xx" can be a two-digit number from 00 to 47, which is associated with Program 10-04-02 by subtracting 1.

This “xx” is also associated with the regular VRS message number (01 to 48), so the user can record the music via phone by using the service code 116 + 7 instead of using the PC.

VRS for MOH Conditions

- When the IntraMail compact flash is installed, this feature cannot be used.
- When the selected music file is not accessible (for example, the file does not exist, the DSP-DBU is not installed, etc.), or all the VRS channels are busy, the system will play Internal MOH (Selection 1) instead.

VRS Enhanced with Call Attendant When Receiving Busy/No Answer at Extension
The function of the VRS has been enhanced. When the system is set up to use VRS greetings for incoming trunks (defined as VRS or DISA in Program 22-02-01), the system now provides an option for the caller to hear another message when no answer or a busy signal is received at the dialed extension. The caller would then be able to call the operator, leave a message, dial another extension, or transfer to a ring group - based on how the options are set in Program 25-06-01.

This option can be set for the system or for each extension.

VoIP

3

Aspire S	Aspire M/L/XL
• Available - 16 IP extensions max.	• Available with software 2.00 or higher. 64 IP extensions maximum with Basic CPU or with Feature Upgrade PAL chip - 512 IP extensions maximum with enhanced CPU.
• Gain Setup for VOIPU available.	• Gain Setup for VOIPU available with software 1.11+.
• Enhanced Echo Adjustment available with software 1.18+.	• Enhanced Echo Adjustment available with software 1.18+.
• Calling Party Number available with software 2.09+.	• Calling Party Number available with software 2.09+.
• Out-of-band DTMF signaling requires software 2.63+.	• Out-of-band DTMF signaling requires software 2.63+.
• SIP requires software 2.63+.	• SIP requires software 2.63+.
• VoIP Port Assignment improved with 4.93+.	• VoIP Port Assignment improved with 4.93+.
• RTP Forwarding available with 4.93+.	• RTP Forwarding available with 4.93+.
• SIP enhancements require software 5.10+.	• SIP enhancements require software 5.10+.
• ICMP Redirect for the VOIPU PCBs require software 5.91+.	• ICMP Redirect for the VOIPU PCBs require software 5.91+.
• RTP Status Change requires software 5.91+.	• RTP Status Change requires software 5.91+.
• Enhanced SIP Terminal functions requires software 5.91+.	• Enhanced SIP Terminal functions requires software 5.91+.
• Carrier B and C mode support requires software 5.92+.	• Carrier B and C mode support requires software 5.92+.

Description

VoIP (voice over Internet protocol or voice over IP) allows the delivery of voice information using the Internet protocol (sending data over the Internet using an IP address). This means that voice information, in a digital form, can be sent in packets over the Internet rather than using the traditional public switch telephone network (CO lines). A major advantage of VoIP and Internet telephony is that it avoids the tolls charged by ordinary telephone service.

Using VoIP equipment at a gateway (a network point that acts as an entrance to another network), the packetized voice transmissions from users within the company are received and routed to other parts of the company's intranet (local area or wide area network) or they can be sent over the Internet using CO lines to another gateway.[1]

The Aspire VoIP supports H.323, H.325, and H.245 trunks and compressions of G.711, G.723.1, and G.729.

IEEE 802.3af Compliance:

- **H.323 IP Telephone (P/N 780005)** - The H.323 phones are not IEEE 802.3af compliant and ***cannot*** use the ILPA adapter for PoE.
- **4-Button Aspire IP Phone (P/N 0890072)** - The Aspire 4-Button IP phones are IEEE 802.3af compliant. *This phone is **not** compatible with NEC PoE or Cisco*.
- **34-Button Aspire IPhone (P/N 0890065B)** - The Aspire IP phones are not IEEE 802.3af compliant unless an ILPA adapter is used for PoE. Refer to the ILPA Adapter Instructions, P/N 0893109 for details.
- **34-Button Aspire IPhone (P/N 0890073)** - The Aspire IP keysets with PoE are IEEE 802.3af compliant. *This phone is **not** compatible with NEC PoE or Cisco*.

Program Available for Gain Setup of VOIPU PCB

The system software provides an option to adjust the gain setting for the VOIPU PCB**.**

Calling Party Number Setup for Trunks and Extensions

The system provides two programs which allow the programmed entry of the Calling Party Number. These entries determine the information displayed when VoIP trunks and extensions are used.

Using LANs

Using a LAN setup (local area network) with the Aspire system complies with the ethernet standard (10Base-T/100Base-TX).

To connect a telephone to a LAN connection, the system allows the use of an Aspire digital IP 34-button keyset (referred to as Aspire IPhone), an Aspire digital keyset with an IP Adapter installed or an H.323 IP digital telephone. For details on installing the IP Adapter, refer to the Aspire Hardware Manual (P/N 0893100).

If connecting a LAN to a WAN (wide area network), follow the instructions included with the ADSL modem or gateway device.

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1. The voice quality of VoIP is dependent on variables such as available bandwidth, network latency and Quality of Service (QoS) initiatives, all of which are controlled by the network and internet service providers. Because these variables are not in NEC's control, it cannot guarantee the performance of the user's IP-based remote voice solution. Therefore, NEC recommends connecting VoIP equipment through a local area network using a Private IP address.

Gatekeeper

With the Aspire system, a separate external gatekeeper is not required unless connecting to an outside H.323 endpoint/gateway which requires an outer gatekeeper or if over 50 outer addresses must be registered. Otherwise, the Aspire provides tables within the system programming for address resolution.

Routers

When purchasing a router for use with the Aspire IP feature, the minimum requirements would be that it provide VPN and QoS. Current VoIP protocols for the Aspire, NGT and H.323 telephones can not communicate over NAT. Therefore, when communications is required over NAT, the router must support VPN. Note that a router which supports ‘VPN Pass Through’ requires a VPN server.

The priority control feature is required to prevent RTP packet loss. If a WAN is used for VoIP only and the bandwidth is wide enough for the VoIP channel, then the QoS feature might not be required.

The following are available routers which provide VPN and QoS:

- NEC IX1000 / 2000 Series
- Yamaha RT105, RTX1000 / 2000
- Cisco 800 Series
- Furukawa FITELnet - F40
- Fujitsu SiR-170, SiR-150

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The following routers provide VPN but no priority control (QoS):

- Linksys BEFSX41, DEFVP41
- OMRON MR104DV, MR104FH
- Allied Telesis AR410

Switches/Hubs for PoE

Using a central power supply or the 8SHUBU PCB with the PoE jumpers set, power over category 5 network cables can be provided. This eliminates the need of installing separate power adapters for each IP phone and it allows for centralized power backup. ***Note: IEEE 802.3af compliance is not available on all Aspire IP phones. Refer to*** **IEEE 802.3af Compliance:** (page 222) ***for details.***

! CAUTION !

Only Aspire IP phones (P/N 0890065) and Aspire IP Adapters (P/N 0890060) and H.323 phones (P/N 780005) must be connected to the 8SHUBU. The provided DC voltage provided through the spare pairs (4/5, 7/8) may damage any other equipment.

If PoE (power over ethernet) is to be used to eliminate the separate power adapters, due to the power requirements, a separate power source is suggested. It is recommended that you use one of the central power supplies below:

- 8SHUBU PCB - Provides 8-port switching hub and the ability to provide PoE for Aspire equipment.
  - 802.1p/1q Support
- 24-port power supply PoE-managed switch (NEC BlueFire 200/24)
  - PoE (Power Over Ethernet) to Aspire IP/H.323 Phones
  - Spare Pair (4/5, 7/8) / Signal Pair (1/2, 3/6) Selection

    For systems which require layer 2 switching capability and PoE, the NEC BlueFire 200/24 switch is recommended. This unit provides layer 2 switch capability in addition to being able to supply ethernet power to 24 NEC IP terminals.
- Cisco Data Switch - CDP Supported

Other manufacturer central power supplies may be usable, but the above items have been tested for compatibility with the Aspire equipment.

Release Link of Media Gateway Channels

When a trunk from Site A rings an extension in Site B, if Site B answers the call and then transfers it back to an extension in Site A, the following occurs:

- On the initial call, there is one media gateway channel used on each site.
- When Site B places the call on hold and transfers it to Site A, another media gateway channel is used in Site B. When Site A answers the call, all the media gateway channels are released.

The release of the media gateway channels is automatic and no programming is required for this operation.

Use of SIP Protocol Available (* Limited Feature Release)

Contact your NEC Sales Representative for details on the current release status of the SIP feature.

SIP (Session Initiation Protocol) is a protocol used for Voice over IP. It is defined by the IETF (Internet Engineering Task Force) in RFC2543 and RFC3261 (RFC3261 requires system software 5.10 or higher). SIP trunking is the term used for linking a PBX, like the Aspire, to the public telephone network by means of VoIP. This provides the possibility for users to place and receive communications and services from any location and for networks to identify the users wherever they are located.

SIP analyzes requests from clients and retrieves responses from servers then sets call parameters at either end of the communication, handles call transfer and termination. The Aspire implementation and programming for SIP and H.323 are very similar. The call routing, call features and speech handling (RTP) are the same - only the signalling protocol is different.

The Aspire system can now support the following:

- IP Trunk : H.323 Trunks / SIP Trunks
- IP Extension : Dterm IP / H.323 Extensions / SIP Extensions

  Support for SIP extensions is the same as H.323 telephones. This SIP extension enhancement requires software 5.91+.
- IP Networking : Networking over H.323

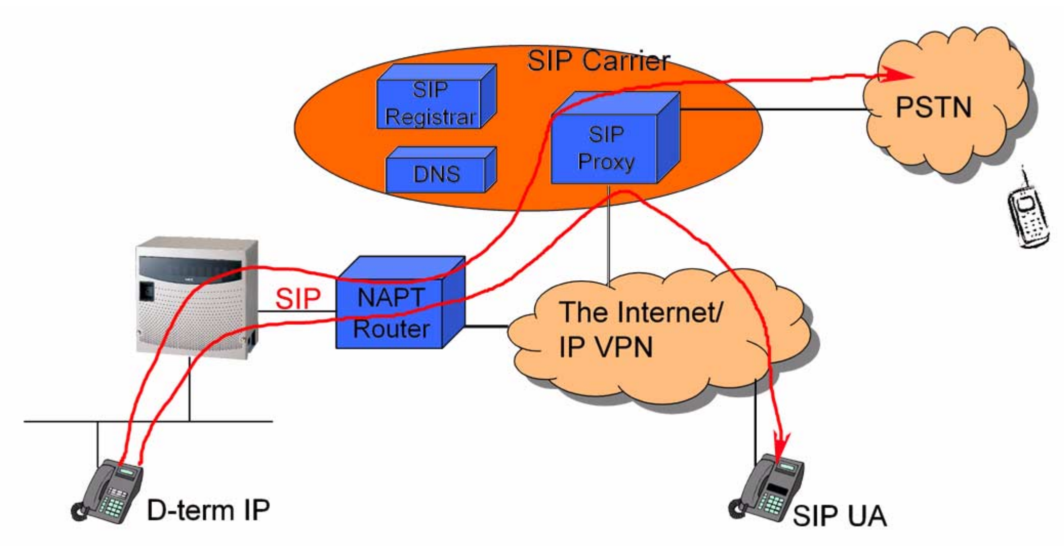
  This protocol can be used simultaneously.

With the Aspire, SIP trunks can receive incoming calls with Caller ID, place outgoing calls, and transfer SIP trunks to IP, SIP, analog and digital stations, and across a network. SIP trunks can be used for making a simple IP networking like H.323 IP trunk inter-connection with Program 10-23.

If a common carrier supports SIP, then the Aspire can connect the SIP Carrier and outgoing calls to the PSTN network and the common IP network using an Aspire SIP trunk.

Currently, however, SIP Centrex Transfer is not supported.

Refer to the Aspire SIP Trunking Manual, P/N 0893214, for details on programming this feature.



Conditions - SIP Trunking

- Aspire does not support a simultaneous using of a SIP trunk inter-connection and a SIP Trunk Carrier connection.
- Aspire supports a "100rel" option and "Session Timer" option.
- Aspire supports "Early Media".
- Aspire supports a "DNS resolution access" and a "IP address direct access" for SIP server.
- Aspire supports a "MD5" Authentication flow for a SIP server.
- Aspire restricts an outgoing call under the following conditions:
  - SIP configuration failed.
  - SIP registration failed.
  - NTCPU/MBU/VOIPU link down.
  - Lack of VOIPU DSP resource.
  - Lack of a VoIP bandwidth.
- Aspire can connect a SIP server over NAPT router by one static global IP address.
- Aspire supports the sub-address feature with SIP trunk inter-connection.
- Aspire supports RFC2833 for DTMF relay feature.
- Aspire SIP does not support T.38 FAX(H.323 trunk supports T.38 FAX).

Conditions - SIP Extensions

- SIP extensions support G.711, G.729 CODEC 20ms

SIP Feature Enhanced for Trunk and Extension Operation

With software 5.10 or higher, the SIP function has been enhanced with the following items:

- **SIP Common**
    - SIP RFC3261 Supported Completely by SIP Trunks (Aspire SIP trunks or SIP Carrier trunks) and SIP Extensions
      The SIP stack has been updated from RFC2543 Base to RFC3261 Base.

- **SIP Trunk Enhancement**
    - ***Support the 401 response for the Initial Invite.***
      If 401 message is sent for the Initial Invite, previously, the system could not respond to the message correctly. This software corrects this problem.
    - ***Support the 401/407 response for the Invite of Session Timer.***
      If 401/407 response is sent for the invite of Session Timer, previously, the system could not send the Invite message with Authentication header. This software corrects this problem.
    - ***Support the 128 byte size of Nonce max value sent by the 401/407 message.***
      Old software could receive the 64 byte size of Nonce max value sent by the 401/407 message. With this change, the Nonce max size is expanded to 128byte.

- **SIP Extension - Hold/Transfer**

  The Aspire will only support SIP extensions which comply with the two standards for the Hold and Transfer features as described in the Internet Engineering Task Force (IETF) documents:
  - draft-ietf-sipping-service-examples-09txt
  - draft-ietf-sipping-cc-transfer-05.txt
  These documents can be obtained at the IEFT web site: www.ietf.org

    - Call Hold
    - Transfer - Screened
    - Protecting Transfer Target
        - When Extension A transfers a call to Extension B, the Caller ID for Extension B can be prevented from showing on Extension A's display.
    - Attended Transfer
        - A transferred call can be rerouted to a different user agent using a GRUU (Globally Routable User Agent URI).
    - Semi-Attended Transfer
        - An unanswered transferred call can route to the extension's voice mail.

Note: If a SIP terminal does a semi-attended Transfer to a trunk, SIP extension and voice mail, the Aspire system calls back to the SIP terminal immediately.

SIP Feature Enhanced for Terminal Operation

The call controller for SIP terminals has been changed with this software from S-Bus based to DECT based. With this change, some functions of the SIP terminal have been enhanced.

Conditions - SIP Trunking

- Aspire does not support a simultaneous using of a SIP trunk inter-connection and a SIP Trunk Carrier connection.
- Aspire supports a "100rel" option and "Session Timer" option.
- Aspire supports "Early Media".
- Aspire supports a "DNS resolution access" and a "IP address direct access" for SIP server.
- Aspire supports a "MD5" Authentication flow for a SIP server.
- Aspire restricts an outgoing call under the following conditions:
  - SIP configuration failed.
  - SIP registration failed.
  - NTCPU/MBU/VOIPU link down.
  - Lack of VOIPU DSP resource.
  - Lack of a VoIP bandwidth.
- Aspire can connect a SIP server over NAPT router by one static global IP address.
- Aspire supports the sub-address feature with SIP trunk inter-connection.
- Aspire supports RFC2833 for DTMF relay feature.
- Aspire SIP does not support T.38 FAX(H.323 trunk supports T.38 FAX).

Conditions - SIP Extensions

- SIP extensions support G.711, G.729 CODEC 20ms

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Volume Controls

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Handset volume reset option requires software 2.63+.	• Handset volume reset option requires software 2.63+.
• Speakerphone volume reset option requires software 5.91+.	• Speakerphone volume reset option requires software 5.91+.

Description

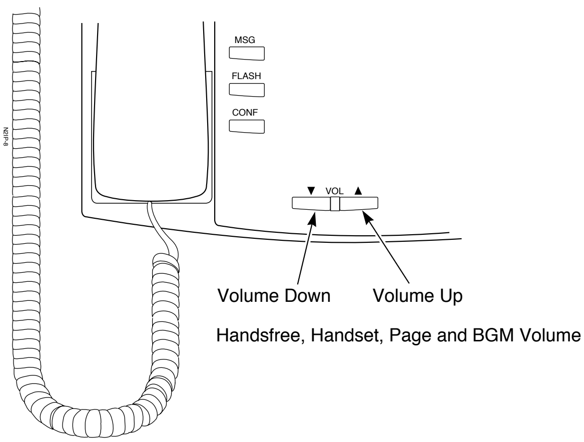
Each keyset user can control the volume of incoming ringing, splash tone, Paging, Background Music, Handsfree and your handset. Keysets consolidate all adjustments into the volume buttons. Pressing the VOLUME ▲ or VOLUME ▼ will adjust the volume level for whichever feature is active (outside call, ICM, ICM ringing, paging, etc.). Pressing these keys when the phone is idle will adjust the contrast level of the telephone’s display. The users should set the volumes for their most comfortable levels.

Handset Volume Reset Option Available

Depending on your software, an option is available which allows the system to either reset the handset volume back to the default setting after hanging up the handset or it can retain the user’s setting.

Option Changed to Retain or Default Speaker Volume Level

Depending on your software level, the volume reset option (**Program 15-02-27 : Multi-Line Telephone Basic Data Setup - Volume of Handset**) may affect the speakerphone volume as well.



Warning Tone For Long Conversation

Aspire S	Aspire M/L/XL
• Available.	• Available.
• Timer Class of Service requires software 5.91+.	• Timer Class of Service requires software 5.91+.

Description

The system can broadcast warning tones to a trunk caller warning them that they have been on the call too long. The tones are just a reminder -- the user may be able to disregard the tones and continue talking if they choose. The outside caller does not hear the warning tones. In addition, warning tones do not occur for Intercom calls and most incoming trunk calls. DISA trunks can also have warning tones. Warning tones are not available to analog single line telephone (SLT) users.

There are two types of warning tones: Alarm Tone 1 and Alarm Tone 2. Alarm Tone 1 is the first set of tones that occur after the user initially places a trunk call. Alarm Tone 2 broadcasts periodically after Alarm Tone 1 as a continued reminder. Each alarm tone consists of three short beeps.

If programmed, DISA calls will be disconnected unless the "continue" code is entered by the user. With the Long Conversation Cutoff feature, incoming or outgoing central office calls can also be disconnected.

Warning Tone for DISA Callers

For DISA callers, with this feature enabled, the warning tone timer begins when an incoming DISA call places an outgoing call and either the inter-digit timer expires or the outgoing call is answered.

With this feature enabled, the warning tone timer begins when an incoming DISA call places an outgoing call and either the inter-digit timer expires or the outgoing call is answered.

If an outside call is transferred to forwarded off-premise using an outside trunk, the warning tone timer begins immediately. This will occur only if either trunk involved in the call is programmed for this feature (Program 14-01-17). When transferring a trunk call off-premise, Program 14-01-13 must be enabled (set to '1').

System Number Plan/Capacities

System Number Plan/Capacities			
System Type:	**Aspire S**	**Aspire M/L**	**Aspire XL**
System			
Analog Caller ID Detector	24	64	64
Classes of Service	15	15	15
Day/Night Mode Numbers	4	8	8
Day/Night Service Patterns	4	32	32
Dial Tone Detector DTMF Receiver	16	64	64
System Ports (trunks and analog/digital extensions)	• ***Software thru 2.21 =*** 8 trunks and 26 extensions • ***Software 2.50+ =*** 8 trunks and 50 extensions	• ***NTCPU with Basic PAL =*** 64 trunks/extensions • ***NTCPU with Feature Upgrade PAL (software 01.00 - 03.10) =*** 64 trunks/extensions • ***NTCPU with Feature Upgrade PAL (software 04.00+) =*** 128 trunks/extensions • ***NTCPU-B*** 200 trunks and 256 extensions	• 200 trunks and 384 extensions
Toll Restriction Classes	15	15	15
Verifiable Account Code Table	2000	2000	2000
Trunk			
Trunk Port Number	1-8	1-200 [1]	1-200
	[1] With the basic NTCPU (P/N 0891002), trunks count toward the total number of allowed hardware ports (64 or 128 ports depending on the PAL EPROM and software installed)		
Trunk Ports (Total) • Analog Trunks • BRI Trunk Ports • T1/PRI Trunk Ports • E&M Analog Trunk Ports • DID Analog Trunk Ports • VoIP Trunk Ports	8 8 4 (8B) N/A N/A 8 8	200 200 96 (192B) 192 60 120 200	200 200 96 (192B) 192 60 120 200
BRIU Logical Ports	T-Bus: 1-8 S-Bus: 1-26	T-Bus: 1-200 S-Bus: 1-256	T-Bus: 1-200 S-Bus: 1-256

System Number Plan/Capacities			
System Type:	**Aspire S**	**Aspire M/L**	**Aspire XL**
COIU: • Physical Ports • Logical Ports	 01-04 0-8	 01-08 0-200	 01-08 0-200
DIOPU: • Physical Ports • Logical Ports	 01-02 LD Trunk: 0-8 OPX: 0-8	 01-08 LD Trunk: 0-200 OPX: 0-25	 01-08 LD Trunk: 0-200 OPX: 0-25
PRIU Logical Ports	N/A	T-Bus: 1-200 S-Bus: 1-256	T-Bus: 1-200 S-Bus: 1-256
TLIU: • Physical Ports • Logical Ports	 N/A	 01-08 0-200	 01-08 0-200
VOIPU: • Physical Ports • Logical Ports	 1-8 0-8	 01-32 0-200	 01-32 0-200
DID Translation Tables	20	20	20
DID Translation Table Entries	2000	2000	2000
DISA • Classes of Service • Users	 15 1-15	 15 1-15	 15 1-15
Ring Groups	1-8	1-100	1-100
Tie Line Classes of Service	N/A	15	15
Tie Line Toll Restriction Classes	N/A	15	15
Trunk Access Maps	1-8	1-200	1-200
Trunk Group Numbers	1-8	1-100	1-100
Trunk Routes	1-8	1-100	1-100

System Number Plan/Capacities

System Type:	Aspire S	Aspire M/L	Aspire XL
Extension			
Telephone Extension Port Numbers • Keysets • Single Line Phones/Analog Devices • VoIP Extensions • Aspire Wireless	1-50 [3] (1-24) (1-18) (1-16) [2, 4] N/A	1-256 (1-256) [2] (1-256) [2] (1-512) [5] 009-512 (manual select) [5] 257-512 (auto select) [5]	1-384 (1-384) (1-384) (1-512) [5] 002-512 (manual select) [5] 385-512 (auto select) [5]
	[2] Counts toward total number of allowed hardware ports (Aspire S=26 ports with software 2.08-2.21 or 50 ports with software 2.50+, Aspire M 64-Port NTCPU=64, Aspire M/L 64-Port NTCPU w/Feature Upgrade PAL and software 4.00+=128, Aspire M/L w/Enhanced NTCPU=256, Aspire XL=384). [3] The total number of ports available is determined by system software. ***Software prior to 2.50*** has 26 ports (24 digital and analog, 2 analog only - a maximum of 16 IP extensions is included in this 26 ports) ***Software 2.50 and higher*** provides 50 ports (Maximum Wired Terminals: 26 Includes keysets, single line telephones, ISDN terminals (APRs NOT included), Maximum IP Terminals: 16 Includes DtermIP and H.323 terminals, Maximum Special Terminals: Reserved for IntraMail: 8 (fixed extension ports 43-50). [4] If the APR-B2 mode is assigned in a system which already has 26 extensions, the number of IP phones is reduced. [5] With the basic NTCPU (P/N 0891002), VoIP and Aspire Wireless extensions count toward the total number of allowed hardware ports (64 or 128 ports depending on the PAL EPROM and software installed). With the Enhanced NTCPU (P/N 0891038), if the number of VoIP and Aspire Wireless phones combined exceeds 256, the number of ports available for keysets or analog devices is then reduced by 1 for each additional IP or Aspire Wireless phone.		
ESIU • Physical Ports • Logical Ports -Tone Ringer (2PGDAD) -Door Box (2PGDAD) -Analog I/F (2PGDAD) -ACI (2PGDAD) -APR for B2 Mode	1-8 1-4 1-4 1-8 1-8 1-42 (descending order)	01-16 1-8 1-8 1-96 1-96 193-256 (descending order) ***With software 4.xx or higher:*** 193-512 (descending order)	01-32 1-8 1-8 1-96 1-96 193-512 (descending order)
SLIU • Physical Ports • Logical Ports	1-8 1-26	01-16 1-256	01-16 1-384
Telephone Extension Number Range			
Virtual Extension Ports	24	256	256
Virtual Extension Port Numbers	01-24	001-256	001-256
Virtual Extension Number Range	Undefined	Undefined	Undefined

System Number Plan/Capacities			
System Type:	**Aspire S**	**Aspire M/L**	**Aspire XL**
2PGDAD Modules	10	56	56
ADA (Recording Jack) Adapters		192	192
Aspire Wireless Bases	N/A	12	12
Aspire Wireless Phones	N/A	120	120
Door Boxes	4	8	8
Door Box Numbers	1-4	1-8	1-8
DSS Consoles Numbers • 24-Button DLS Consoles, Maximum Installed • 110-Button DSS Consoles, Maximum Installed	8 24 4	8 256 32	8 384 32
Handsfree Adapter (HF-R)	24	192	192
Operator Access Number	0	0	0
Operator Extension	1	1	1
Ringdown Assignments	512	512	512
SLT Adapters		16	16
Voice Mail Master Numbers			
Abbreviated Dialing			
Abbreviated Dialing Groups	8	64	64
Abbreviated Dialing Bins	0-1999	0-1999	0-1999
Abbreviated Dialing Table-Common	1000	1000	1000
ACD			
ACD Groups	N/A	64	64
ACD Agent Extensions	N/A	512	512
ACI			
ACI Groups	4	16	16
ACI Ports	8	96	96
Automated Attendant			
VRS Message Numbers	1-48	1-48	1-48
Conference			
Conference Circuits	32 - maximum (32 Parties Per Conference)	64 - maximum (32 Parties Per Conference)	64 - maximum (32 Parties Per Conference)

System Number Plan/Capacities			
System Type:	**Aspire S**	**Aspire M/L**	**Aspire XL**
Data Communication Interfaces			
APR Software Port Numbers	193-256 ***With software 4.xx or higher:*** 193-512	193-256 ***With software 4.xx or higher:*** 193-512	193-512
APA Adapters	24	192	192
APR Adapters	B1 = 24 B2 = 8 prior to 2.50 or 16 with 2.50+	192	192
CTA or CTU Adapters	24	128	128
Module Extension Number Range			
Department and Pickup Groups			
Department (Extension) Group Numbers	1-8	1-64	1-64
Department (Extension) Group Number Range			
Call Pickup Group Numbers	1-8	1-64	1-64
Hotline			
Internal Hotline	512	512	512
External Hotline	512	512	512
Paging and Park			
Internal Page Group Numbers	0, 1-8	0, 1-9 or 01-64	0, 1-9 or 01-64
External Page Group Numbers	0, 1-8	0, 1-8	0, 1-8
External Speakers • NTCPU • PGDAD Module	8 N/A (1-8)	9 (1) (1-8)	9 (1) (1-8)
Park Group Numbers	1-64	1-64	1-64
Park Orbits	1-64	1-64	1-64
SMDR			
SMDR Ports	1-2	1-8	1-8
VRS			
VRS (on DSP Daughter Board)	1	1	1
VRS Channels	8	16	16
VRS Attendant Messages	3	3	3
VRS Recordable Messages	48	48	48

System Number Plan/Capacities			
System Type:	**Aspire S**	**Aspire M/L**	**Aspire XL**
Voice Mail			
Ports for IntraMail	8 (fixed extension ports 43-50)	N/A	N/A
Ports for External Voice Mail	72 [6]	72	72
	[6] Though this is the maximum available in the NVM-Series voice mail, as each voice mail port requires an analog port, the total number is restricted by the Aspire S system to a maximum of 16.		
VoIP			
VoIP Extensions	16 [4]	• ***NTCPU with Basic or Feature Upgrade PAL (software 01.00 - 03.10)*** = 64 • ***NTCPU with Feature Upgrade PAL (software 04.00+)*** = 128 • ***NTCPU-B*** = 512	512
	[4] If the APR-B2 mode is assigned in a system which already has 26 extensions, the number of IP phones is reduced.		
ADA2 (Recording Jack) Adapters		192	192
IP Adapters		256	256
IP Phones	16	512	512
PSA (Power Failure) Adapters		192	192
RAS Unicast Ports	0-65535	0-65535	0-65535
Call Signaling Ports	0-65535	0-65535	0-65535
NGT Signal Receive Ports	0-65535	0-65535	0-65535
DRS Ports	0-65535	0-65535	0-65535
RTP Ports	0-65535	0-65535	0-65535
RTCP Ports	0-65535	0-65535	0-65535
H.245 Ports	0-65535	0-65535	0-65535
DSP Resources	01-32	01-32	01-32
H.323 Alias Addresses	1-6	1-6	1-6

System Number Plan/Capacities			
System Type:	**Aspire S**	**Aspire M/L**	**Aspire XL**
Passwords			
User Password for setting Toll Restriction Override and Changing Class of Service using a service code	0000	0000	0000
Programming Passwords			
Level 1 (MF) PCPro/WebPro User Name:	374772 NEC-I	374772 NEC-I	374772 NEC-I
Level 2 (IN) PCPro/WebPro User Name:	12345678 ASPIRE	12345678 ASPIRE	12345678 ASPIRE
Level 3 (SA) PCPro/WebPro User Name:	0000 ADMIN1	0000 ADMIN1	0000 ADMIN1
Level 4 (SB) PCPro/WebPro User Name:	9999 ADMIN2	9999 ADMIN2	9999 ADMIN2
Programming Password Users	8	8	8
Note:			
Extension numbers can be three or four digits long. See Flexible System Numbering.			

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Aspire S System Specifications

Aspire S System Capacities		
Cabinets	1	
Analog Trunks (CO/PBX lines)	8	
Digital Key Telephones Analog Single Line Telephones IP Telephones Virtual Extensions	24* 18* 16 * 24	* combined total to 26 (includes extension ports provided on the CPU)
24-Button DLS Consoles 110-Button DSS Consoles	24 4	1 maximum per extension 4 maximum per extension.
Conference Circuits	32 (32-parties max per Conference)	
ADA Adapter	24	installs on a keyset
ADA2 Adapter	16	installs on an IP keyset
APA Adapters	24	installs on a keyset
APR Adapters	B1 = 24 B2 = 8 prior to software 2.50 / 16 with software 2.50+	installs on a keyset
CT-11 Cordless Headset Telephone	5	start with 3 - increase if no interference experienced
CTA Adapter	24	installs on a keyset
CTU Adapter	24	installs on a keyset
Handsfree Adapter (HF-R)	24	installs on a keyset
IP Adapter	16	installs on a keyset
IP Power Failure Adapter (PSA)	16	installs on an IP keyset
Power Failure Telephones	2	provided by COIU PCBs
SLT Adapter	8	
2PGDAD Modules	10 (4 for ACI ports, 4 for Door Boxes, 4 for Pages)	combined total to 10
Door Box/Door Unlock Contacts Internal Page Zones External Page Zones	4 8 8	
Dial Tone Detector Circuits DTMF Receiver Circuits	16* 16*	combined total to 16
Universal PCB slots	6	
* ***NOTE:*** Maximum capacities above are determined by maximum PCB configuration allowed. When installing single line sets or DISA, CPU circuits must be allocated for DTMF receivers. To install single line sets with CO/PBX line access, CPU circuits must be allocated for dial tone detection.		

Aspire S PCB Capacities	
CPU Central Processing Unit	1
ENTU LAN CTI	1
8ESIU 8 Digital Stations	2
4SLIU 4 Analog Stations	4
4SLIDB 4 Analog Stations Daughter Board (installs on 4SLIU)	0 - If 4 4SLIU PCBs installed 1 - If 3 4SLIU PCBs installed 2 - If 2 4SLIU PCBs installed
4COIU 4 Analog/Loop Start Trunks (no ground start)	2
2DIOPU 2 DID/OPX Trunks	4
4VOIPU 4 VoIP Media Gateway	3
4VOIPDB 4 VoIP Media Gateway Daughter Board (installs on 4VOIPU)	3

Aspire M/L System Specifications

Aspire M/L System Capacities	NTCPU-A with Basic PAL Chip	NTCPU-A with Feature Upgrade PAL Chip (Software 1.00-3.07)	NTCPU-A with Feature Upgrade PAL Chip (Software 4.00+)	NTCPU-B	Notes
Cabinets	1	2 (Main and 1 Expansion Cabinet)	2 (Main and 1 Expansion Cabinet)	2 (Main and 1 Expansion Cabinet)	
Power Supplies	2	4 (2 per cabinet)	4 (2 per cabinet)	4 (2 per cabinet)	
Trunks (CO/PBX lines)	64 [1]	64 [1]	128 [2]	200 [3]	[1] combined trunk and extension total of 64 [2] combined trunk and extension total of 128 [3] combined trunk and extension total of 256
Digital Key Telephones Analog Single Line Telephones Aspire Wireless 2.4 GHz Telephones IP Telephones	64 [1] 64 [1] 64 [1] 64 [1]	64 [1] 64 [1] 64 [1] 64 [1]	128 [2] 128 [2] 120 [2] 128 [2]	256 [3] 256 [3] 120 [4] 512 [5]	[1] combined trunk and extension total of 64 [2] combined trunk and extension total of 128 [3] combined trunk and extension total of 256 [4] in addition to the digital or analog ports [5] combined total, including extensions and trunks, cannot exceed 512 - over 256 IP extensions will use 1 digital port (reducing the max. for digital extensions)

Aspire XL System Specifications

Aspire XL System Capacities (requires software 4.xx or higher)	NTCPU-A with Basic PAL Chip	NTCPU-A with Feature Upgrade PAL Chip	NTCPU-B	Notes
Cabinets	1	2 (Main and 1 Expansion Cabinet)	2 (Main and 1 Expansion Cabinet)	
AC/DC Power Supplies	2	2	2	
DC/DC Converters	1	2 (1 per cabinet)	2 (1 per cabinet)	
Trunks (CO/PBX lines)	64 [1]	128 [2]	200 [3]	[1] combined trunk and extension total of 64 [2] combined trunk and extension total of 128 [3] combined trunk and extension total of 384
Digital Key Telephones Analog Single Line Telephones Aspire Wireless 2.4 GHz Telephones IP Telephones	64 [1] 64 [1] 64 [1] 64 [1]	128 [2] 128 [2] 120 [2] 128 [2]	384 [3] 384 [3] 120 [4] 512 [5]	[1] combined trunk and extension total of 64 [2] combined trunk and extension total of 128 [3] combined trunk and extension total of 384 [4] in addition to the digital or analog ports [5] combined total, including extensions and trunks, cannot exceed 512 - over 256 IP extensions will use 1 digital port (reducing the max. for digital extensions)

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Aspire M/L/XL System Specifications

Aspire M/L/XL System Capacities	NTCPU-A with Basic PAL Chip	NTCPU-A with Feature Upgrade PAL Chip (any version of software)	NTCPU-B	Notes
24-Button DLS Consoles 110-Button DSS Consoles	64 32	64 32	256 32	1 maximum per extension 4 maximum per extension.
Conference Circuits	64 (32-parties max per Conference)	64 (32-parties max per Conference)	64 (32-parties max per Conference)	
ADA Adapter	64	64	192	installs on a keyset
ADA2 Adapter	64	64	192	installs on a keyset
APA Adapter	64	64	192	installs on a keyset
APR Adapter	64	64	192	installs on a keyset
CT-11 Cordless Headset Telephone	5	5	5	start with 3 - increase if no interference experienced
CTA Adapter	64	64	128	installs on a keyset
CTU Adapter	64	64	128	installs on a keyset
Handsfree Adapter (HF-R)	64	64	192	installs on a keyset
IP Adapter	64	64	256	installs on a keyset
PSA Adapter	64	64	256	installs on a keyset
Power Failure Telephones	14	28	28	provided by COIU PCBs
SLT Adapter	16	16	16	
2PGDAD Modules	56 (48 for ACI ports, 4 for Door Boxes, 4 for Pages)	56 (48 for ACI ports, 4 for Door Boxes, 4 for Pages)	56 (48 for ACI ports, 4 for Door Boxes, 4 for Pages)	
Door Box/Door Unlock Contacts Internal Page Zones External Page Zones	8 64 9	8 64 9	8 64 9	
Dial Tone Detector Circuits DTMF Receiver Circuits	32* 32*	64** 64**	64** 64**	* combined total to 32 ** combined total to 64
Universal PCB slots Main Cabinet Expansion Cabinet	1-8 (Not including NTCPU slot)	1-8 (Not including NTCPU slot) 8-16 (Not including EXIFU slot)	1-8 (Not including NTCPU slot) 8-16 (Not including EXIFU slot)	

* ***NOTE:*** Maximum capacities above are determined by maximum PCB configuration allowed. When installing single line sets, DISA, or tie lines, NTCPU circuits must be allocated for DTMF receivers. To install single line sets with CO/PBX line access, or when installing immediate-start tie lines, NTCPU circuits must be allocated for dial tone detection.

Aspire M/L/XL PCB Capacities	NTCPU-A with Basic PAL Chip	NTCPU-A with Feature Upgrade PAL Chip (Software 1.00-3.07)	NTCPU-A with Feature Upgrade PAL Chip (Software 4.00+)	NTCPU-B
NTCPU Central Processing Unit	1	1	1	1
EXIFU Expansion PCB	-	1	1	1
DSPDB Resource/VRS Daughter Board (installs on NTCPU)	-	1	1	1
8ESIU 8 Digital Stations	8	8	16	16
16ESIU16 Digital Stations	4	4	8	16
32ESIU 32 Digital Stations	2	4	4	***With P/N 0891000 power supplies (Aspire M/L) :*** 4 ***Software 4.xx+ and P/N 0892011/0892012 power supplies (Aspire XL):*** 12 (8 max. per cabinet)
8SLIU 8 Analog Stations	7	7	15	15
8SLIDB 8 Analog Stations Daughter Board (installs on 8SLIU)	3	3	8	15
16DSTU 16 i-Series Keyset Interface	3	3	7	15
4COIU-LS1 4 Analog/Loop Start Trunks (no ground start)	7	14	15	15
8COIU-LS1 8 Analog/Loop Start Trunks (no ground start)	7	7	15	15
4COIU-LG1 4 Analog/Loop Start Trunks (with ground start)	7	14	15	15
8COIU-LG1 8 Analog/Loop Start Trunks (with ground start)	7	7	15	15
4DSIU Aspire Wireless Interface	1	1	1	1
8DSIU Aspire Wireless Interface	1	1	1	1
12DSIU Aspire Wireless Interface	1	1	1	1
2BRIU 2 Two-Channel BRI Circuits	-	14	15	15
4BRIU 4 Two-Channel BRI Circuits	-	7	15	15
8BRIU 8 Two-Channel BRI Circuits	-	3	As T-Bus: 8 As S-Bus: 7	As T-Bus: 12 As S-Bus: 15
1PRIU 24 T1/PRI Trunks / Channels	-	2	5	8

Aspire M/L/XL PCB Capacities

	NTCPU-A with Basic PAL Chip	NTCPU-A with Feature Upgrade PAL Chip (Software 1.00-3.07)	NTCPU-A with Feature Upgrade PAL Chip (Software 4.00+)	NTCPU-B
4TLIU 4 E&M Tie Line Trunks	-	14	15	15
4DIOPU 4 DID/OPX Trunks	7	14	15	15
8DIOPU 8 DID/OPX Trunks	7	7	15	15
2FMSU 2 Flash Memory Voice Mail	1	1	1	1
4FMSU 4 Flash Memory Voice Mail	1	1	1	1
4VMSU 4 HDD Voice Mail	1	1	1	1
4VMDB 4 HDD Voice Mail - Daughter Board	1	1	1	1
8SHUBU 8 Switch Hub	4	8 (4 per cabinet)	8 (4 per cabinet)	8 (4 per cabinet)
4VOIPU 4 VoIP Media Gateway	8 *	16 *	16 *	16 *
4VOIPDB 4 VoIP Media Gateway Daughter Board (installs on 4VOIPU or 16VOIPU)	8 *	8 *	16 *	16 *
16VOIPU 16 VoIP Media Gateway	-	4 *	8 *	16 *
16VOIPDB 16 VoIP Media Gateway Daughter Board (installs on 4VOIPU or 16VOIPU)	-	3 *	6 *	16 *
16CNF-U20 16 Conference Bridge	3	7	15	15

* In addition to the load factor, additional factors may limit these quantities further (such as available bandwidth, VIF and compression).

Environmental Requirements

Meeting established environmental standards maximizes the life of the system. Refer to the Standard Practices Manual for further information. Be sure that the site is not:
1. In direct sunlight or in hot, cold or humid places.
2. In dusty areas or in areas where sulfuric gases are produced.
3. In places where shocks or vibrations are frequent or strong.
4. In places where water or other fluids comes in contact with the main equipment.
5. In areas near high-frequency machines or electric welders.
6. Near computers, telexes, microwaves, air conditioners, etc.
7. Near radio antennas (including shortwave).

Environmental Specifications

Cabinets, PCBs and Key Telephones
Temperature: 0ºC - 40ºC (32 - 104ºF)
Humidity: 10-90% RH
Door Box
Temperature: -20ºC - 60ºC (-4 - 140ºF)
Humidity: 20-80% (non-condensing)
Aspire S Power Supply
Operating:
Temperature: 0ºC - 60ºC (32 - 140ºF)
Humidity: 10-90% RH
Storage:
Temperature: -20ºC - 75ºC (-4 - 167ºF)
Humidity: 10-95% RH
Aspire Power Supply
Operating:
Temperature: 0ºC - 40ºC (32 - 104ºF)
Humidity: 20-90% RH
Storage:
Temperature: -40ºC - 75ºC (-4 - 167ºF)
Humidity: 10-95% RH
VMSU-A1 PCB
Temperature: +5ºC - 40ºC (41-104ºF)
Humidity: 10-90% RH

Power Requirements

A dedicated 110 VAC 60 Hz circuit located within seven feet of the cabinet is required. You should install a separate dedicated outlet for each cabinet.

Caution
Double Pole/Neutral Fusing
(power supply fuses located at both the L and N side)

Site Requirements

The system can be floor-, wall- or rack-mounted. Brackets secure each cabinet to a wall.

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Aspire S Electrical Specifications

Power Supply

AC Power Supply
Dedicated 15 Amp circuit
Power Requirements: 120 VAC @ 15A KSU
Power Consumption (max.): 65W
Input Voltage: 85VAC to 138VAC
Rated frequency: 50/60 Hz
Input Frequency: 47 - 63 Hz
Phase and Wire: Single, 2-Wire
Grounding Requirements: No. 14 AWG copper wire
With input voltage of 120 VAC and with full load conditions:
Output Power (Watts max.): 80W
AC Input I: 1.5A
VA @ 120V: 180VA
KWh @ AC Input I x 120V/1000: 0.18 KWh
BTU (KWh x 3413): 614 btu

Caution
Double Pole/Neutral Fusing
(power supply fuses located at both the L and N side)

Output Voltage Types:	+3.3VDC (-3%, +1%)	+5VDC (+/- 2%)	-28VDC (+/- 5%)
Output Current 1	2.0A - 5.0A	1.2A - 1.6A	2.4A - 1.5A
Output Current 2	2.0A - 5.0A	1.2A - 1.6A	2.4A - 1.5A
Ripple/Noise	150mV p-p	150mV p-p	150mV p-p
Overvoltage Protection	3.7 - 5.0V	5.6 - 7.5V	31.5 - 40V
Overcurrent Protection	6.0 - 9.0A	1.9 - 3.5A	5.1 - 8.7A

Aspire M/L Electrical Specifications

Power Supply

AC Power Supply
Dedicated 15 Amp circuit
Power Requirements: 120 VAC @ 15A Main Cabinet
Power Consumption: Base Cabinet=360W (493 VA), Expansion Cabinet=360W (493 VA), total 720W (986 VA)
Input Voltage: 85VAC to 135VAC
Frequency: 47 Hz - 63Hz (Rated frequency: 50/60 Hz)
Phase and Wire: Single, 2-Wire
Grounding Requirements: No. 14 AWG copper wire

With input voltage of 120 VAC and with full load conditions:

Output Power: One Power Supply: Base Cabinet=100.5W
Two Power Supplies: Base Cabinet=200.9W, Expansion Cabinet=200.9, total 401.8W
AC Input I: One Power Supply: Base Cabinet=2.06A
Two Power Supplies: Base Cabinet=4.11A, Expansion Cabinet=4.11A, total 8.22A
VA @ 120V: One Power Supply: Base Cabinet=246.6VA
Two Power Supplies: Base Cabinet=493.2 VA, Expansion Cabinet=493.2 VA, total 986.4 VA
KWh @ AC Input I x 120V/1000: One Power Supply: Base Cabinet=0.247 KWh
Two Power Supplies: Base Cabinet=0.493 KWh, Expansion Cabinet=0.493 KWh, total 0.986 KWh
BTU (KWh x 3413): One Power Supply: Base Cabinet=842 btu
Two Power Supplies: Base Cabinet=1683 btu, Expansion Cabinet=1683 btu, total 3366 btu

Caution
Double Pole/Neutral Fusing
(power supply fuses located at both the L and N side)

Output Voltage Types	+3.42VDC (-3%, +1%)	+5VDC (+/- 2%)	-48VDC (+/- 5%)
Output Current 1	0.0A - 6.0A	0.0A - 5.0A	0.0A - 1.0A
Output Current 2	0.0A - 3.2A	0.0A - 1.0A	0.0A - 2.0A
Ripple/Noise	50mV p-p	100mV p-p	200mV p-p
Overvoltage Protection	3.7 - 8.0V	5.6 - 13.0V	-55.0 - 64.8V
Overcurrent Protection	6.6 - 7.7A	5.5 - 6.5A	2.2 - 2.6A

Aspire XL Electrical Specifications

Power Supply
AC Power Supply
Dedicated 15 Amp circuit
Input Voltage: 85VAC to 135VAC
Frequency: 50/60 Hz
Rated Input: 100Vac / 120Vac / 220-240Vac
7.6A / 6.1A / 3.3A-3.1A

Main Output Terminal	+3.3V	+5V	-48V
Output Voltage	+3.42V	+5.0V	-51.90V
Output Voltage Range	+3.317V - +3.454V (+3.42V-3%+1%)	+4.90V - +5.10V (+5.0V±2%)	-49.84V - -52.94V (-51.90V+2%-4%)
Maximum Load Current	12.0A	10.0A	8.0A
Load Regulation Range	0.0A~12.0A	0.0A~10.0A	0.0A~8.0A
Ripple/Noise	100mV p-p or less	150mV p-p or less	200mV p-p or less
Psophometric Noise	-65dBm or less	-65dBm or less	-65dBm or less
Overvoltage Protection	latched off	latched off	latched off
Overcurrent Protection	latched off	latched off	auto-recovered (after removal of fault)
Capacitive Load	12000 μ F	12000 μ F	24000 μ F

Aspire Soft Phone Requirements

Before setting up *Aspire Soft Phone*, ***make sure the VoIP feature is programmed and operating*** in the Aspire system. Refer to the Aspire Software Manual (P/N 0893200) for details.

Conditions

- Aspire Soft Phone does not support G.723.
- Using DHCP, the IP address of the NTCPU cannot be obtained.

For the *Aspire Soft Phone* application, please confirm the following requirements are met for the PC.

Minimum PC Requirements
Aspire Soft Phone is installed on a PC. Please confirm that the PC meets the minimum PC requirements before installing the *Aspire Soft Phone*.

Internet Explorer version 6.0 or higher is required. If an older version of Internet Explorer is installed, the *Aspire Soft Phone* installation is stopped and the installation cannot be completed correctly.

Required Environment	
CPU	Intel Pentium® III, Celeron™ Processor 600MHz or higher AMD Athlon™, Duron™ Processor 700MHz or higher
Memory	128MB or more
HDD	30MB HDD empty space
Sound	Sound equipment on Windows® operating system
Video	SVGA (800x600) display resolution and high color (16 bit, 65536 colors) video card and monitor
Peripheral Equipment	Speaker and MIC (or Headset)
OS	Microsoft® Windows® 2000 Professional Microsoft® Windows® XP Home Edition Microsoft® Windows® XP Professional
Font Size	Small size (Microsoft® Windows® 2000 Professional) Normal size (Microsoft® Windows® XP Professional, Microsoft® Windows® XP Home Edition)
Browser	Microsoft® Internet Explorer 6.0 or higher

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CallAnalyst Requirements

Ultra Call Analyst
Minimum PC Requirements
- PC with Pentium Processor
- 256 MB RAM
- VGA monitor 800 x 600 resolution (recommended SVGA 1024 x 768)
- Windows Operating Software - 95/98/ME, NT-SP 3 or later, 2000, XP
- 500 MB of free hard drive space
- CD-ROM drive (for software installation)
- Available serial port and RS-232 cable
- Printer (if required to print reports)

CallAnalyst Enterprise Server

Main Server Minimum PC Requirements
Hardware
- PC with Pentium 4 Processor
- 512 MB RAM
- SVGA Monitor with 1024 x 768 resolution
- 2 GB of free hard drive space
- CD-ROM drive (for software installation)
- Available serial port and RS-232 cable (if required)
- Network Interface Card (NIC)
- Printer (if required to print reports)

Software
- Windows NT 4.0 (workstation or server) w/SP6, 2000 Professional w/SP3, XP Professional w/ SP1 or 2003 Server
- MS SQL Server 2000 or MSDE (Microsoft Database Engine) for the database (MSDE is included on the application CD)
- Microsoft Internet Explorer 5.0 or higher (Internet Explorer is included on the application CD)

Network Client / Remote Site Reporting Client Minimum PC Requirements
Hardware
- PC with Pentium III Processor
- 256 MB RAM
- 1 GB of free hard drive space
- CD-ROM drive (for software installation)
- Network Interface Card (NIC)
- Available serial port and RS-232 cable (if required)

Software
- Windows NT 4.0 w/ SP6, or 2000 Professional w/ SP3, XP Professional w/ SP1 or 2003 Server
- Microsoft Internet Explorer 5.0 or higher (Internet Explorer is included on the application CD)

inDepth Requirements

inDepth Server Requirements		
	Minimum	**Recommended**
Computer/Processor	Pentium 3 700 Megahertz (MHz) or equivalent 256 RAM	Pentium 4 2.4 GHz or above 512 RAM
Operating System	Windows XP Professional * Windows 2000 Server Windows 2003 Server	Windows XP Professional * Windows 2000 Professional, SR-3 Windows 2000 Server Windows 2003 Server
	* Windows XP Home Edition not not allow for clients.	
Hard Disk Free Space	Hard disk space requirements will vary depending on the operating system configuration and call traffic:	
	2 GB	10 GB
Graphics Card	256 color resolution 800 x 600 (SVGA)	256 color resolution 800 x 600 (SVGA) or above
Video Monitor	14”	17” or above
CD-ROM Drive	Installed	Installed
Network Card	10 Mbps	100 Mbps
Uninterruptable Power Supply	No	Yes
Additional Software: Internet Explorer	Version 4	Version 6
Backup	N/A	CD-R, Tape or Network
Serial Ports	Serial port requirements dependant upon configuration.	
Events from Switch Physical Wallboard External Modem	1 1 1	
Parallel Ports (Free)	1	
Microsoft Network	Required	
TCP/IP	- Each PC must have TCP/IP installed. - Each inDepth PC must have a unique IP address - If running a networked inDepth system, each inDepth. PC must be able to communications with each other inDepth PC via TCP/IP. - It is recommended that name resolution be implemented.	
Static IP Address	Required	
Network File Sharing	Enabled	
Remote Support Software Connection	PC Anywhere Version 9 or Above Direct Telephone Line or Remote Access Server	

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inDepth Requirements (cont'd)		
inDepth Client - Sub-Supervisor Requirements		
	Minimum	**Recommended**
Computer/Processor	Pentium 3 700 Megahertz (MHz) or equivalent 256 RAM	Pentium 4 2.4 GHz or above 512 RAM
Operating System	Windows XP Professional * Windows 2000 Server Windows 2003 Server	Windows XP Professional * Windows 2000 Professional, SR-3 Windows 2000 Server Windows 2003 Server
	* Windows XP Home Edition not not allow for clients.	
Hard Disk Free Space	30 MB	30 MB
Graphics Card	256 color resolution 800 x 600 (SVGA)	256 color resolution 800 x 600 (SVGA) or above
Video Monitor	14”	17” or above
Network Card	10 Mbps	100 Mbps
Uninterruptable Power Supply	No	Yes
Additional Software: Internet Explorer	Version 4	Version 6
Installation	CD-ROM or Network Drive	CD-ROM or Network Drive
Serial Ports	Serial port requirements dependant upon configuration.	
Physical Wallboard	1	
Microsoft Network	Required	
TCP/IP	Installed	

***** The above requirements are based on systems with no other applications installed. If you run other applications, you may need to increase processor speed, memory and hard disk space accordingly.

inDepth Requirements (cont'd)

inDepth Client - inView LAN Wallboard Requirements

	Minimum	Recommended
Computer/Processor	Pentium 4 2.4 GHz or above, 512 RAM	
Operating System	• Windows XP Home Edition • Windows XP Professional • Windows 2000 Professional SR-3 • Windows 2000 Server • Windows 2003 Server	
Hard Disk Free Space	30 MB	30 MB
Graphics Card	256 color resolution 800 x 600 (SVGA)	
Video Monitor	14"	
Network Card	10 Mbps	
Uninterruptable Power Supply	No	
Additional Software: Internet Explorer	Version 4	
Installation	CD-ROM or Network Drive	
Microsoft Network	Required	
TCP/IP	Installed	

* The above requirements are based on systems with no other applications installed. If you run other applications, you may need to increase processor speed, memory and hard disk space accordingly.

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Aspire VoIP Requirements		
Category	**Feature**	**Notes**
IP Address	DHCP Server	NTCPU
	DHCP Client	VOIPU PCB or IP Phone
QoS	802.1p/1q	
	L3 QoS (ToS)	Diffserv/IP Precedence
Maintenance	HTTP Server	NTCPU
Server	H.323 Gatekeeper	For H.323 Phone Registration and Routing
VLAN	Tag and port-based VLAN	
VoCoder	G.711 μ-law/A-law	
	G.729a	
	G.723.1	
	Fax Relay	
Jitter Buffer Size	Set by system programming	
RTP Length	Set by system programming	
Echo Canceller Tail Size	Set by system programming	
Level Adjustment	Set by system programming	
Protocol	H.323	
	NGT (Next Generation Telephone)	
IP Phone	H.323 Phone	H.323 Phone
	NGT Phone	Maximum 512 Phones
IP Trunk	H.323 Trunk	Maximum 200 Trunks

Aspire 4-Button IP Phone Specifications	
Display	3x24 Character LCD
Line Appearance	4 Line Keys/Programmable Function Keys
10/100M Ethernet Port	1 Port
Compression Support	G711/G729
Speakerphone	Full Duplex Handsfree
Multiple Power Options	PoE (IEEE802.3af) Local: AC Adapter Option (AC-R or AC-2R, P/N 780135) ***The phone is not compatible with NEC PoE or Cisco Data Switch - CDP Supported.***
Power Consumption	7 watts per phone (146.3mA @ 48V)
QoS	IEEE802.1p/q, DiffServ
Downloadable Software	Using TFTP/FTP
DHCP	Supported
Message Wait Lamp	Yes
Feature Keys	9 Keys plus Volume Keys
Wall Mount Option	Yes-Built-In
EMC	Class B
Soft Keys	4 with Check/Clear Keys
Body Color	Black

Aspire 34-Button PoE IP Phone Specifications	
Display	3x24 Alpha/Numeric Character LCD
Fixed Function Keys	11
Programmable Function Keys	24
Programmable One-Touch Keys	10
Soft Keys	4 Keys with Check/Clear Keys
Speaker Phone	Yes (Full Duplex)
Message Waiting Lamp	Yes
Headset Interface	Yes
LAN Interface	10Base-T/100Base-TX, Full/Half, Auto-Negotiate/Fixed, 2 Port
Voice CODEC	G.711, G729a
IP Address Setting	DHCP, Direct Setting
QoS	ToS (IP Precedence, Diffserv)
VLAN	TagVLAN (IEEE802.1Q/p)
Multiple Power Options (Center Power Feeding)	PoE (IEEE802.3af) Local: AC Adapter Option (AC-R or AC-2R, P/N 780135) ***The phone is not compatible with NEC PoE or Cisco Data Switch - CDP Supported.***
Power Consumption	7 watts per phone (146.3mA @ 48V)
Body Color	Black
Power Consumption	4.2W

Aspire 34-Button IP Phone Specifications	
Display	3x24 Alpha/Numeric Character LCD
Fixed Function Keys	11
Programmable Function Keys	24
Programmable One-Touch Keys	10
Soft Keys	4 Keys with Check/Clear Keys
Speaker Phone	Yes (Full Duplex)
Message Waiting Lamp	Yes
Headset Interface	Yes
LAN Interface	10Base-T/100Base-TX, Full/Half, Auto-Negotiate/Fixed, 2 Port
Voice CODEC	G.711, G729a
IP Address Setting	DHCP, Direct Setting
QoS	ToS (IEEE 802.1 p/q Based and DiffServ)
VLAN	TagVLAN (IEEE 802.1 p/q)
Multiple Power Options (Center Power Feeding)	PoE (IEEE 802.1 p/q) using Aspire 8SHUBU PCB Local: AC Adapter Option (AC-R or AC-2R, P/N 780135) ***The phone is compatible with NEC PoE or Cisco Data Switch - CDP Supported.***
Body Color	Black
Power Consumption	4.2W

H.323 IP Phone Specifications	
Display	2x24 Character LCD
Line Appearance	2 Line Keys
10/100M Ethernet Port	1 Port
Compression Support	G711/G729
Speakerphone	Full Duplex Handsfree
Multiple Power Options	PoE (IEEE 802.1 p/q) using Aspire 8SHUBU PCB Local: AC Adapter Option (AC-R or AC-2R, P/N 780135) ***The phone is compatible with NEC PoE or Cisco Data Switch - CDP Supported.***
QoS	IEEE802.1p/q, DiffServ
Downloadable Software	Using TFTP/FTP
DHCP	Supported
Message Wait Lamp	No
Feature Keys	20 Keys including Volume Keys
Wall Mount Option	Yes-Built-In
Soft Keys	No
Body Color	Black

Mechanical Specifications

Equipment	Width	Depth	Height	Weight
Aspire S KSU Cabinet	16 7/16”	5 3/8”	13 3/16”	13.22 lbs fully equipped
Aspire M/L/XL KSU Cabinet	16 1/2”	10”	15 1/2”	55 lbs 11 oz fully equipped
Aspire XL Power Supply Cabinet	16 1/2”	10”	15 1/2”	45 lbs 12 oz fully equipped
2-Button Telephone	6 3/8”	8 3/4”	2 3/4”	2 lbs 1 oz
22-Button Non-Display Keyset	7 3/4”	9 2/8”	3 7/8” no leg extension 5 7/8” legs fully extended	2 lbs 4 oz
22-Button Display Keyset	7 3/4”	9 2/8”	3 7/8” no leg extension 5 7/8” legs fully extended	2 lbs 4 oz
34-Button Display Keyset	7 3/4”	9 2/8”	3 7/8” no leg extension 5 7/8” legs fully extended	2 lbs 5 oz
Super Display Keyset	7 3/4”	10 7/8”	3 7/8” no leg extension 6 1/8” legs fully extended, display down 7 7/8”legs fully extended, display up	2 lbs 7 oz
4-Button IP Keyset	7 1/2”	8 1/2”	5 3/8”	2 lbs 15 oz
34-Button IP Keyset - P/N 0890065	7 3/4”	9 2/8”	3 7/8” no leg extension 5 7/8” legs fully extended	2 lbs 13 oz
34-Button IP Keyset - P/N 0890072B	7 3/4”	9 2/8”	3 7/8” no leg extension 5 7/8” legs fully extended	2 lbs 13 oz
110-Button DSS	7 3/4”	9 2/8”	3 3/16”	1 lb 12 oz
24-Button DLS	2”	9 2/8”	2 15/16”	12 oz
Door Box	3 7/8”	1”	5 1/8”	6.5 oz

2PGDAD Module/NTCPU Input/Output

Audio/Music Input	
Input Impedance:	47 KOhm @ 1Khz
Audio/Paging Output	
Output Impedance:	600 Ohms @ 1 KHz
Maximum Output:	+3 dBm
Relay Contacts	
Configuration:	Normally Open
Maximum Contact Ratings:	24 VDC, 0.5A 120 VAC, 0.25A
Night Mode Relay Connection, Input	
Break:	48 VDC
Make:	7 mA

Adapter Inputs

ADA	+27VDC, -48VDC -28VDC ⎓ 620 Ohms (dip switch setting on adapter)
ADA2	+27VDC, -48VDC -28VDC ⎓ 620 Ohms (dip switch setting on adapter)
APR	+27/24VDC ⎓
CTA	+27VDC, -48VDC -28VDC ⎓
IP	+27VDC, -48VDC

BGM/MOH Music Source Input

Input Impedance:	47KOhm / 1Khz
Input Level:	Nominal 250 mV (-10 dBm)
Maximum Input:	1V RMS

Inputs for MOH and BGM are located on the NTCPU PCB. The 2PGDAD also provides MOH inputs.

Door Box/External Paging

Output Impedance:	600 Ohm
Output Level:	Nominal 250 mV (-10 dBm)
Maximum Output:	400 mV RMS
Configuration:	Normally open

LAN Specifications

Standard	IEEE802.3 10Base-T and 100Base-TX Compliant
Access	CSMA/CD
Capacity	10Base-T/100Base-TX; Aspire S: 1 Port for LAN Terminal Aspire: 8 Ports for LAN Terminal
I/F (Layer 1)	- Speed; 10Mbps/100Mpbs Auto Negotiation - Cable; Category 5 or better, Straight/Cross Cable Auto Crossover
Aspire Switching	- Store and Forward Layer 2 Switching - MAC Address Auto Recognition - Store Max. 1,000 MAC Addresses - Flow control in Back Pressure Mode Compliant

Single Line Telephones

This section applies to the following telephones:

DTH-1-1 (P/N 780034) - DTR-1-1 (P/N 780020, 780021) - DTR-1HM-1 (780025 or 780026)

Message Waiting Lamp

Lamp On (Activation Voltage): 88-108 VDC

Lamp Off (Deactivation Voltage): 53 VDC or less

SLT Adapter

Constant Current Circuit: Current fixed at 47 mA

Signal Method

On-Hook Condition: 48VDC
Ringer Signal: 180 Vp-p, 16Hz

SLIU PCB / SLIDB

Aspire S Signal Method

On-Hook Condition: -28VDC +/- 1.4VDC
Message Waiting Signal: -90VDC +/- 4.5VDC
Ringer Signal: 65Vrms, 20Hz (with no load)

Note: The Message Waiting lamp is not provided to the analog ports on the CPU.

Aspire Signal Method

On-Hook Condition: -46VDC +/- 3VDC (with no load), nominal 90 VDC
Message Waiting Signal: -112VDC +/- 3VDC or FSK
Ringer Signal: 75Vrms +-1Vrms (no load condition), 20Hz +-1%

UL Listed System

A label will be affixed to the product with the letters UL inside a circle which is the symbol used by UL to indicate that a product is UL Listed. If you see a small “c” outside the symbol, then the product also meets the requirements for Canada.

Recommended Usage Guidelines for Cordless Phones
In ideal conditions, multiple spread spectrum type cordless phones (Cordless II can be utilized in the same environment. However, due to the possible interference problems caused by the bases being placed in close proximity to each other, we recommend the following:
Spread Spectrum Phones (Cordless II P /N 730088) Where users require greater range on the cordless phones and 3 or less cordless phones are being used at a specific site, we recommend using the spread spectrum cordless phone.
FM Modulation Phones (Cordless Lite II P /N 730087) Where more than 3 cordless phones are to be used at one specific site, we recommend using the FM modulation cordless phones which have 30-channel capability.
CT-11 Codless Headset Phone (P /N 730090) It is recommended to start to 3 phones, and if no interference is experienced, you can increase the capacity to a recommended maximum of 5.
Dterm Headset Cordless II Phone (P /N 730091) The recommended maximum range is between 15-50. This depends on how often the units are used and the proximity of the users. Frequency hopping is used to automatically change channels if interference is present. If the phone detects wideband interference (multiple users in a small area), the range will be decreased until an adequate link is formed. The present of other 900 MHz devices will cause interference and may need to be removed.
Note: The range of the phones depends largely on the environmental factors, such as the building structure, the size of the room, RF interference and other electronic equipment installed in the same area. For optimum range and performance, we suggest the following: • Place the base units at least 15 feet apart. The performance of the phones become more stable when the distance between the bases is greater. • Place the base unit in the center of the coverage area. • If the phone will also be used in an outdoor area, like a parking lot, install the base unit in an area close to the window. • If a phone experiences interference and noise, press the channel key to select another channel.

Cordless Lite II	
General	
Frequency Control	Phase Lock Loop
Modulation	900 MHz Narrow Band FM with ADPMC (digital)
Operating Temperature	0ºC - +50ºC (+32ºF to +122ºF)
Bandwidth	50 kHz
Data Transmission Speed	688 bps
Channels	30 Channels
Base Unit	
Receive/Transmit Frequency	902 MHz - 928 MHz
Power Requirements	10V DC from supplied AC adapter
Size	4 1/4” W x 7 5/8” D x 2 1/4” H
Weight	Approximately 13.7 oz.
Handset	
Receive/Transmit Frequency	902 MHz - 928 MHz
Power Requirements	NiMH Battery
Size	2” W x 1 1/4” D x 5 1/2” H (with antenna)
Weight	8.7 oz. with battery
Battery	Capacity 700 mAh, 3.6V
Talk Mode	5 hours (typical)
Standby Mode	40 hours (typical)
Specifications shown are typical and subject to change without notice.	
Battery Charger	
Power Requirements	9V DC from supplied AC adapter
Size	1 3/8” W x 1 1/2” D x 2 1/4” H
Range	350’ (depending on environmental conditions)

Cordless II	
General	
Frequency Control	Phase Lock Loop
Modulation	Digital Spread Spectrum
Operating Temperature	0ºC - +50ºC (+32ºF to +122ºF)
Output Power	60 mW
Occupied Bandwidth	$\pm$ 500 KHz
Data Transmission Speed	688 bps
Channels	10 Channels
Base Unit	
Receive/Transmit Frequency	902 MHz - 928 MHz
Power Requirements	10V DC from supplied AC adapter
Size	4 1/4” W x 7 5/8” D x 2 1/4” H
Weight	11.8 oz.
Handset	
Receive/Transmit Frequency	902 MHz - 928 MHz
Power Requirements	Nickel-Cadmium Battery Pack
Size	2 1/4” W x 1 1/2” D x 6 5/16” H
Weight	8.6 oz.
Battery	Capacity 800 mAh, 3.6V
Charging Time	5-6 hours max. to full charge
Talk Mode	6 hours (typical)
Standby Mode	4 days (typical)
Specifications shown are typical and subject to change without notice.	
Battery Charger	
Power Requirements	9V DC from supplied AC adapter
Size	4 3/4” W x 4 1/2” D x 3 1/2” H
Weight	6.5 oz.
Range	350’ (depending on environmental conditions)

FCC Registration Information					
Manufacturer: FCC Part 15 Registration:			NEC Infrontia, Inc. Class A		
Model: FCC Registration Number: (Refer to the label on the System Cabinet for the FCC Registration Number.			**Aspire S** KF: US:NIFKF06BASPIRES MF: US:NIFMF06BASPIRES PF: US:NIFPF06BASPIRES		
Reg. Status	**Facility Interface Code (FIC)**	**Mfrs. Port Identifier**	**Ringer Eq. Number**	**Service Order Code (SOC)**	**Network Jacks**
Original	02LS2, 02IS2	4COIU	0.6B	9.0F, 6.0P	RJ21X
Model: FCC Registration Number: (Refer to the label on the Main Cabinet for the FCC Registration Number.			**Aspire** KF: US:NIFKF07BASPIRE MF: US:NIFMF07BASPIRE PF: US:NIFPF07BASPIRE		
Reg. Status	**Facility Interface Code (FIC)**	**Mfrs. Port Identifier**	**Ringer Eq. Number**	**Service Order Code (SOC)**	**Network Jacks**
Original	02LS2	4COIU-LS1	0.7B	9.0F	RJ21X
Original	02LS2	8COIU-LS1	0.7B	9.0F	RJ21X
Original	02GS2	4COIU-LG1	0.7B	9.0F	RJ21X
Original	02GS2	8COIU-LG1	0.7B	9.0F	RJ21X
Original	02RV2-T	4DIOPU	---	AS.2	RJ21X
Original	02RV2-T	8DIOPU	---	AS.2	RJ21X
Reg. Status	**Analog Private Line Interfaces**	**Mfrs. Port Identifier**		**Service Order Code (SOC)**	**Network Jacks**
Original	TL11M	4TLIU	---	9.0F	RJ2EX

FCC DID Requirements

Federal Communications Commission DID Requirements
This equipment must operate in a manner that is not in violation of Part 68 rules. This equipment returns answer supervision to the Public Switched Network when the DID trunk is: (1) answered by the called station; (2) answered by the attendant; (3) routed to a recorded announcement that can be administered by the CPE user; (4) routed to a dial prompt.

The equipment returns answer supervision on all DID calls forwarded back to the Public Switched Telephone Network except when: (1) a call is unanswered; (2) a busy tone is received; (3) a reorder tone is received.

When ordering DID Service, provide the telco with the following information:

Aspire S FCC Registration Number:	
KF	US:NIFKF06BASPIRES
MF	US:NIFMF06BASPIRES
PF	US:NIFPF06BASPIRES
DID Facility Interface Code:	TBD02RV2-T
DID Service Order Code:	TBD9.0F
DID Answer Supervision Code	TBDA S.2
DID USOC Jack Type:	RJ21X
Aspire FCC Registration Number:	
KF	US:NIFKF07BASPIRE
MF	US:NIFMF07BASPIRE
PF	US:NIFPF07BASPIRE
DID Facility Interface Code:	02RV2-T
DID Service Order Code:	9.0F
DID Answer Supervision Code	A S.2
DID USOC Jack Type:	RJ21X

Please note the following:
1. DID services must be purchased from the local telephone company.
2. Refer to the Software Manual for detailed DID description, conditions, and programming instructions.

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Cabling Requirements

1. Do not run station cable in parallel with the AC source, telex or computer, etc. If the cables are near cable runs to those devices, use shielded cable with grounded shields or install the cable in conduit.
2. When cables must be run on the floor, use cable protectors.
3. Cable runs for key telephones, single line telephones, Door Boxes, CTA or CTU adapters, and 2PGDAD Modules must be a dedicated, isolated cable pair.
4. The Telco RJ21X and cross-connect blocks should install to the right of the Main Cabinet. Extension blocks and cross-connect blocks should be installed to the left of the Main Cabinet.

Aspire S Cable Requirements

Device	Cable Type	Cable Run Length (ft)	Notes
Key Telephone, DSS Console:	ICT 2-wire 26 AWG ICT 2-wire 24 AWG ICT 2-wire 22 AWG	656 (200 meters) 984 (300 meters) 1640 (500 meters)	
Single Line Telephone, Analog Terminals:	ICT 2-wire 26 AWG ICT 2-wire 24 AWG ICT 2-wire 22 AWG 900 ohm loop resistance	6,561 (2000 meters) 10,498 (3200 meters) 17,388 (5300 meters)	Includes resistance of telephone
SLT Adapter: SLTAD to SLT:	ICT 2-wire 26 AWG ICT 2-wire 24 AWG ICT 2-wire 22 AWG 500 ohm loop resistance (including phone's DC termination)	656 (200 meters) 984 (300 meters) 1640 (500 meters)	
2PGDAD Adapter:	ICT 2-wire 26 AWG ICT 2-wire 24 AWG ICT 2-wire 22 AWG	656 (200 meters) 984 (300 meters) 1640 (500 meters)	
Door Box to 2PGDAD:	ICT 2-wire 26 AWG ICT 2-wire 24 AWG ICT 2-wire 22 AWG 200 ohm loop resistance (excluding internal resis-tance of Door Box)	656 (200 meters) 984 (300 meters) 1640 (500 meters)	
Using Under Carpet Cable: The cable is not twisted, but flat parallel wire. Good impedance balance cannot be expected and reduces the cable length to no more than 300m when using the flat cable.			
PC to CTA	Serial Straight Thru Cable	49.21 (15 meters	D-Sub 9-Pin Female (PC) to D-Sub 9-Pin Male (CTA)
CTA to Printer	Serial straight cable	49.21 (15 meters)	D-Sub 9-Pin Female (Printer) to D-Sub 9-Pin Male (CTA)

Aspire S Cable Requirements

Device	Cable Type	Cable Run Length (ft)	Notes
CPU to PC:	Serial cross cable	49.21 (15 meters)	D-Sub 9-Pin Female (PC) to D-Sub 9-Pin Male (CPU)
	4-wire LAN (UTP) cable with category 3 or higher for 10Base-T, and category 5 or higher for 100 Base-TX Ethernet cross cable	328.08 (100 meters)	
CPU to Switching Hub:	4-wire LAN (UTP) cable with category 3 or higher for 10Base-T, and category 5 or higher for 100 Base-TX Ethernet straight cable	328.08 (100 meters)	
CPU to Printer:	Serial cross cable	49.21 (15 meters)	D-Sub 9-Pin Female (Printer) to D-Sub 9-Pin Male (CPU)
IP Telephone	LAN (UTP) cable with category 3 or higher for 10Base-T, and category 5 or higher for 100 Base-TX	328.08 (100 meters)	Regardless of whether it receives power from the AC adapter or switching hub.
VOIPU PCB to IP Telephone	Standard Ethernet Cable (cross cable)	328 (100 meters)	
VOIPU PCB to Router	Standard Ethernet Cable (straight cable)	328 (100 meters)	
PSA Adapter	ICT 2-wire	4,547 (1,386 meters)	Includes DC resistance of outside line - excludes internal resistance of IP telephone
ENTU PCB to PC	Standard Ethernet Cable (cross-cable)	328 (100 meters)	
ENTU to Switching Hub	Standard Ethernet Cable (straight-cable)	328 (100 meters)	
DID Trunks:	ICT 2-wire 24 AWG Loop resistance (DP only) 9842 ohm @35mA 500 ohm @20 mA	20,997 (6400 meters)	Includes SLT or exchange Includes resistance of telephone
2BRIU to ISDN Terminals:	ICT 4-wire 24 AWG	328 (100 meters) with Point-Multipoint short connection 984 (300 meters)300m with Point-Multipoint long connection 1,640 (500 meters) with Point-Point connection	

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Aspire M/L/XL Cable Requirements

Device	Cable Type	Cable Run Length (ft)	Notes
Key Telephone, DSS Console:	2-wire 26 AWG 2-wire 24 AWG 2-wire 22 AWG	1312 1968 2624	
Single Line Telephone, Analog Terminals:	2-wire 26 AWG 2-wire 24 AWG 2-wire 22 AWG	13,123 20,997 34,776	
20mA Setting: 35mA Setting:	1,500 ohm loop resistance 900 ohm loop resistance		Includes resistance of telephone
SLT Adapter:	2-wire 26 AWG 2-wire 24 AWG 2-wire 22 AWG	1312 1968 2624	
SLTAD to SLT:		500 ohm (including phone's DC termination)	
2PGDAD Adapter:	2-wire 26 AWG 2-wire 24 AWG 2-wire 22 AWG	1312 1968 2624	
Door Box to 2PGDAD:	2-wire 26 AWG 2-wire 24 AWG 2-wire 22 AWG	200 328 550	
Using Under Carpet Cable: The cable is not twisted, but flat parallel wire. Good impedance balance cannot be expected and reduces the cable length to no more than 300m when using the flat cable.			
NTCPU to PC:	Serial cross cable (null modem) Ethernet cross cable USB cable (USB1.1)	49.21 (15 meters) 328.08 (100 meters) 16.40 (5 meters)	
NTCPU to Hub:	Ethernet straight cable	328.08 (100 meters)	
NTCPU to Printer:	Serial cross cable	49.21 (15 meters)	
IP Telephone	LAN (UTP) cable with category 3 or more for 10Base-T, and 5 or more for 100 Base-TX	328.08 (100 meters)	Regardless of whether it receives power from the AC adapter or switching hub.
PC to CTA	Serial Straight Thru Cable	4.92	
16VOIPU PCB to IP Telephone and Router	Standard Ethernet Cable	328	
DID Trunks:		DTMF = 1,500 ohm DP = 3,000 ohm	Includes SLT or exchange
8SLIU/8SLIDB to Analog Terminals:		Loop resistance 900 ohm @35mA, 1500 ohm @20 mA	

Aspire M/L/XL Cable Requirements			
Device	**Cable Type**	**Cable Run Length (ft)**	**Notes**
2/4/8BRIU to ISDN Terminals:	4-wire 24 AWG	100m with Point-Multi-point short connection 300m with Point-Multi-point long connection 500m with Point-Point connection	
1PRIU to ISDN Terminal:	4-wire 24 AWG	164.02 (50 meters)	
ILPA Adapter	Cross-over cable		

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Aspire S Configuration Guide

System Cabinet								
Slot Assignment								
CPU	8-Channel Digital/Analog Slots							
	Digital Ports 1-8	Analog Ports 1-2						
	1	2	3	4	5	6	7	8

* Slots 1 and 2 are fixed - these are located on the CPU and are part of the basic system.

Aspire M/L/XL Configuration Guide

Expansion Cabinet								
Power Supply Type								
☐ Qty 1 - P/N 0891000			☐ Qty 2 - P/N 0891000			☐ Qty 1 - P/N 0892011 and P/N 0892012		
Slot Assignment								
EXIFU Expansion Interface	32-Channel Digital/Analog Slots							
	1	2	3	4	5	6	7	8

Main Cabinet								
Power Supply Type								
☐ Qty 1 - P/N 0891000			☐ Qty 2 - P/N 0891000			☐ Qty 1 - P/N 0892011 and P/N 0892012		
Slot Assignment								
NTCPU	32-Channel Digital/Analog Slots							
	1	2	3	4	5	6	7	8

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Parts List

Station Equipment	
Description	**Part Number**
2-Button Telephone - Black	0890047
2-Button Telephone - White	0890048
22-Button Handsfree Non-Display Telephone - Black	0890041
22-Button Handsfree Non-Display Telephone - White	0890042
22-Button Handsfree Display Telephone - Black	0890043
22-Button Handsfree Display Telephone - White	0890044
34-Button Handsfree Display Telephone - Black	0890045
34-Button Handsfree Display Telephone - White	0890046
34-Button Super Display Telephone - Black	0890049
34-Button Super Display Telephone - White	0890050
4-Button Aspire IP Phone - Black (IEEE 802.3af PoE compliant)	0890072
34-Button Aspire IPhone - Black (IEEE 802.1p/1q support - **not** IEEE 802.3af PoE compliant)	0890065
34-Button Aspire IPhone - Black (IEEE 802.3af PoE compliant)	0890073
24-Button DLS Console - Black	0890053
24-Button DLS Console - White	0890054
110-Button DSS Console - Black	0890051
110-Button DSS Console - White	0890052
NEC H.323 IP Telephone	780005
NEC H.323 IP Telephone AC Adapter	780600
Analog (Corded) Telephones:	
DTR-1-1 (BK) Single Line Telephone with redial, flash, mw lamp, data jack ringer/handset volume control - Black	780020
DTR-1-1 (WH) Single Line Telephone with redial, flash, mw lamp, data jack ringer/handset volume control - White	780021
DTR-1HM-1 (BK). Same as P/N 780020 with 8 programmable feature/speed dial keys - Black	780025
DTR-1HM-1 (WH). Same as 780021 with 8 programmable feature/ speed dial keys - White	780026
DTH-1-1 (BK) Single line telephone with 4 programmable feature/ speed dial keys, mw lamp, redial, flash & mute keys	780034
Conference Max:	
Conference Max	750073
Conference Max Plus	750074
Conference Max Expansion Kit (requires Conference Max P/N 750073)	750156
Conference Max Plus Pod Charger (spare power supply charger)	750152
Conference Max Plus Pod Spare Battery Pack	750153
Conference Max to Base Spare RJ45 CAT 5 12’ Cable	750157
Conference Max to Conference Max Spare RJ45 CAT 5 25’ Cable	750158

Station Equipment

Description	Part Number
Cordless Telephones and Options:	
Analog Cordless Phone SLT DTR-1R-2 (BK) (5.8 GHz)	730093
NEC Cordless II Telephone (Uses 1 Aspire digital station port)	730088
NEC Cordless II - Charger Unit	730621
NEC Cordless II - Belt Clip	730620
NEC Cordless II - Cordless II Leather Case	730626
NEC Cordless II - Replacement Battery	730622
NEC Cordless Lite II Telephone (Uses 1 Aspire digital station port)	730087
NEC Cordless Lite II - Replacement Battery	730631
NEC Cordless Lite II - Wall Mount Unit for Charger	730633
NEC Cordless Lite II - Belt Clip	730634
NEC Cordless Lite II - Charger Unit	730632
AC for Charger (Cordless II or Cordless Lite II)	730619
AC for Base (Cordless II or Cordless Lite II)	730618
Headset (Cordless II or Cordless Lite II)	730602
Wall Mount Plate for Base	730608
Aspire Wireless Telephone and Options:	
Base Station	780136
Handset with Battery	780004
Leather Cover with Clip	780148
Charging Cradle	780137
Repeater Unit	780138
External Antenna with Cable for Repeater	780145
AC Adapter	780139
Battery	780140
Handset Belt Clip	780141
Repeater Programming Kit	780142
Service Tool Kit	780143
Deployment Tool	780144
2.4G Deployment/Diagnostic Kit	780146
Special Additional Handset for Diagnostic Kit	780147

Peripheral Station Equipment

Description	Part Number
2PGDAD Module (for Door Box/Page/ACI)	0891027
AC Adapter (AC-2R)	780135
Analog Interface with Ringing Adapter (APR) (Requires P/N 780135 AC-2R Adapter)	0890056
Analog Interface without Ringing Adapter (APA)	0890057
Aspire Telephone Trainer	0890071
Call Recording Adapter (ADA)	0890055
Door Box	922450

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Peripheral Station Equipment (cont'd)	
Description	**Part Number**
Headsets (Modular):	
Plantronics Dterm Headset Cordless II (for APR)	730091
Plantronics CS-50 USB Headset	730092
Polaris™ SupraPlus Binaural, Noise Cancelling	750645
Polaris™ SupraPlus Monaural, Noise Cancelling	750643
Polaris™ SupraPlus Monaural, Voice Tube	750644
Polaris™ Encore Monaural - Voice Tube	750634
Polaris™ Encore® Binaural, Noise Cancelling	750635
Polaris Tristar - Voice Tube	750630
Polaris Mirage - Voice Tube	750631
CT-11 Cordless Headset Telephone	730089
CT-11 Cordless Headset Telephone with M175 Headsest	730090
Headsets (Pin-Type):	
Headset for Cordless Telephones (MX-150) - used with 730090	750642
NEC Headset for Codless Telephones (M175) - used with 730090	750637
Polaris Headset Accessories:	
Ear Cushion (Pkg of 2)	750656
Clothing Clip	750657
Wind Screen	750650
Clear Voice Tube for Encore and Tristar	750652
Clear Voice Tube for Mirage and Supra	750651
Polaris Extension Cable (10')	750655
Rainbow Voice Tube Pack for Encore and Tristar (Pkg of 6)	750654
Rainbow Voice Tube Pack for Mirage and Supra (Pkg of 6)	750653
Spare/Replacement AC Adapter for 730091	750658
Spare/Replacement Battery for 730091	750659
IP Adapter (IP)	0890060
In-Line Power Adapter (ILPA)	780122
NEC Audio Emcee:	
Audio Emcee	750316
Replacement Audio Emcee Remote	750317
Replacement RCA Cable	750318
Power Failure Adapter for IP Telephone (PSA)	0890067
Recording Adapter for IP Telephone (ADA2)	0890066
RS-232C Adapter (CTA)	0890058
SLT Adapter	0891026
Speakerphone Adapter - Black (Requires P/N 780135 AC-2R Adapter)	0890062
Speakerphone Adapter - White (Requires P/N 780135 AC-2R Adapter)	0890063
USB Adapter (CTU) (Requires P/N 780135 AC-2R Adapter)	0890059
Voice Security Recorder (VSR) with USB Cable and Software	780275
Voice Security Recorder (VSR) Manager Software	780274

Aspire Applications	
Description	**Part Number**
CTI Applications	
PC Attendant Console Software on CD	0891086
TAPI 1.x License (1)	0891101
TAPI 1.x Licenses (5)	0891100
inDepth/inDepth Plus - *Aspire M/L/XL Only*	
Aspire inDepth Lite Software (Dongle and Software)	0892144
Aspire inDepth Software (Dongle and Software)	0892102
Aspire inDepth+ Software (Dongle and Software)	0892103
Aspire inDepth Supervisor	0892125
Aspire inDepth Lite to inDepth Upgrade	0892141
Aspire inDepth Lite to inDepth Plus Upgrade	0892142
inDepth to inDepth Plus Upgrade	0892143
inDepth Upgrade i-Series to Aspire	0892153
Aspire inView for 5 Agents	0892126
Aspire inView for 10 Agents	0892127
Aspire inView for 15 Agents	0892128
Aspire inView for 20 Agents	0892129
Aspire inView for 25 Agents	0892130
Aspire inView for 30 Agents	0892131
Aspire inView for 40 Agents	0892132
Aspire inView for 50 Agents	0892133
Aspire inView Upgrade to Add 5 Agents	0892134
Aspire inView Upgrade to Add 10 Agents	0892135
Aspire inView Upgrade to Add 15 Agents	0892136
Aspire inView Upgrade to Add 20 Agents	0892137
Aspire inView Upgrade to Add 25 Agents	0892138
Aspire inView Upgrade to Add 30 Agents	0892139
Aspire inView Upgrade to Add 40 Agents	0892140
Aspire inDepth/inDepth+ Demo CD	0892101
Soft Phone	
Soft Phone Software on CD	0892014
Video Soft Phone Software on CD	0892015
NEC USB **Handset** - UTR-1-1	780097
NEC USB **Handset** - UTR-1-1RS (RoHS Compliant)	780597
NEC USB Phone B/BU **Handset**	780094
NEC USB **Headset** - DSP 300	750639
NEC USB **Headset** - DSP 400	750640
Plantronics CS-50 USB **Headset**	730092

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Aspire Applications (cont'd)	
Ultra CallAnalyst	
Ultra CallAnalyst Lite	0891081
Ultra CallAnalyst Lite - to - Full Upgrade	0891082
Ultra CallAnalyst Full	0891083
Ultra CallAnalyst Full + 1 Client	0891084
Ultra CallAnalyst Additional 1 Client	0891085
Ultra CallAnalyst Enterprise Additional Remote Site License Requires 0891095.	0891094
Ultra CallAnalyst Enterprise Server & 120 Ports	0891095
Ultra CallAnalyst Enterprise Remote Site License (with no Ports) Requires 0891095.	0891096
Ultra CallAnalyst Enterprise Additional Port License Requires 0891095.	0891097
Ultra CallAnalyst Enterprise Additional Network Client License Requires 0891095.	0891098
Ultra CallAnalyst Enterprise Site Reporting Thin Client Requires 0891095.	0891099
Ultra CallAnalyst Software Upgrade (for any CallAnalyst software)	0891080

Aspire S Common Equipment

Description	Part Number
Aspire S 8 Slot KSU	0890005
Floor/Desk Stand Mounting Bracket	0891303
Mod 8 to 25 Pair (Unterminated) Installation Cable	808920
32 MB Compact Flash Card with System Software	0891065
32 MB Compact Flash Card - Blank	0891064
VRS Compact Flash Card	0891040
IntraMail - 4 Port, 8 Hour Kit	0892180
IntraMail - 8 Port, 16 Hour Kit	0892182
Null Modem Cable for PCPro	0892004
CPU Battery	EX0254-0040
56K Modem	858620
Aspire S Demo Kit with 4 Port, 8 Hour IntraMail	0892006
Soft-Sided Aspire S Demo Case with Wheels	0893500

Aspire S PCBs

Description	Part Number
Common Cards	
Ethernet Option Daughter Board	0891053
DSP Resource Daughter Board	0891003
Trunk Interfaces	
4 CO Loop Start Trunk PCB	0891046
2 DID/OPX PCB	0891047
2 BRI PCB - *Check with your NEC Sales Representative for availability.*	0891050
Station Interfaces	
8 Aspire S Digital Station PCB	0891051
4 Analog Station PCB	0891048
4 Analog Station Expansion Daughter Board	0891049
4 Channel VoIP Media Gateway PCB	0891054
4 Channel VoIP Media Gateway Expansion Daughter Board	0891055

Aspire S System Bundles

Description	Part Number
Aspire S/NVM-2e - 2P Bundle	0892031
Consists of:	
1 - 0890005 Aspire S KSU	
1 - 0891046 4 CO Loop Start Card	
4 - 0890043 22B Display Tel (BK)	
1 - 17780-2P - NVM-2e 2 Port VM	
Aspire S/NVM-2e - 4P Bundle	0892032
Consists of:	
1 - 0890005 Aspire S KSU	
1 - 0891046 4 CO Loop Start Card	
1 - 0891048 4 Analog Station Card	
4 - 0890043 22B Display Tel (BK)	
1 - 17780-4P - NVM-2e 4 Port VM	

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Aspire M/L/XL Common Equipment	
Description	**Part Number**
Aspire 8 Slot KSU	0890000
Power Supply - Aspire M/L	0891000
Power Supply Set - Includes:	0890069
AC/DC Power Supply - Aspire XL	0892011
DC/DC Converter - Aspire XL	0892012
Power Supply Cable for Power - Aspire XL	0892013
Power Supply Cable for Signal - Aspire XL	0892010
Power Supply Cabinet - Aspire XL	0890068
19” Rack Mount Bracket	0891300
Null Modem Cable (9-pin, 6’ F-F) for Aspire Mail/PCPro	0892004
NTCPU Battery	EX054-0040
Mod 8 to 25 Pair (Unterminated) Installation Cable	808920
KSU Expansion Set	0891001
64 Port Basic CPU	0891002
256/384 Port CPU	0891038
Feature Upgrade PAL (PAL-A) Chip for 0891002	0891039
32MB Flash Memory Compact Flash Card PCMCIA Card Adapter - with System Software	0891060
32MB Flash Memory Compact Flash Card PCMCIA Card Adapter - Blank	0891061
DSP Resource Daughter Board	0891003
VRS Compact Flash Card	0891040
56K Modem	858620
Conference Bridge (16 Ports) PCB	0891069
Kentrox Satellite 932 CSU	859451

Aspire M/L/XL Voice Mail	
Description	**Part Number**
Aspire Mail 2-to-4 Port Upgrade (for 0891032)	0891044
Aspire Mail 4 Port Expansion Daughter Board (only used on 0891037)	0891045
Aspire Mail 4 Port-30 Hour Voice Mail	0891052
Aspire Mail 4 Port Expansion License (for use with 0891052)	0891066
Aspire Mail DMS 4 Port-130 Hour Voice Mail with Desktop Messaging/Call Control	0891030
Aspire Mail DMS 4 Port Expansion License (only used on 0891030)	0891010
Aspire Mail DMS 8 Port-130 Hour Voice Mail with Desktop Messaging/Call Control	0891031
Aspire Mail DMS 8 Port Expansion Daughter Board (only used on 0891030 or 0891031)	0891036
Aspire Mail DMS 10 Seat Expansion License	0891020
Aspire Mail Plus 4 Port Expansion Daughter Board (only used on 0891033)	0891034
Aspire Mail Plus 8 Port Expansion Daughter Board (only used on 0891056)	0891057
Aspire Mail Plus AMIS Networking Activation	0891059
IntraMail - 4 Port, 8 Hour Kit	0892180
IntraMail - 8 Port, 16 Hour Kit	0892182
Null Modem Cable (9-pin, 6’ F-F) for Aspire Mail/PCPro	0892004

Aspire M/L/XL PCBs

Description	Part Number
Trunk Interfaces	
4 CO Loop Start Trunk PCB	0891005
8 CO Loop Start Trunk PCB	0891004
4 CO Loop Start/Ground Start Trunk PCB	0891029
8 CO Loop Start/Ground Start Trunk PCB	0891028
4 DID/OPX PCB	0891013
8 DID/OPX PCB	0891012
4 E&M Tie Line PCB	0891011
2 BRI PCB	0891006
4 BRI PCB	0891007
8 BRI PCB	0891008
PRI/T1 PCB	0891009
Station Interfaces	
8 Aspire Digital Station PCB	0891015
16 Aspire Digital Station PCB	0891014
32 Aspire Digital Station PB	0891058
16 i-Series Digital Station PCB	0891016
8 Analog Station PCB	0891017
8 Analog Station Expansion Daughter Board	0891018
4 Aspire Wireless PCB	0891090
8 Aspire Wireless PCB	0891091
12 Aspire Wireless PCB	0891092
10/100Base-T 8-Port Switching Hub (8SHUBU) PCB	0891021
4 Channel VoIP Media Gateway PCB	0891042
4 Channel VoIP Media Gateway Expansion Daughter Board	0891043
16 Channel VoIP Media Gateway PCB (B-Series)	0891019
16 Channel VoIP Media Gateway Expansion Daughter Board (A-Series) (for 0891042 or 0891022)	0891023
16 Channel VoIP Media Gateway Expansion Daughter Board (B-Series) (only for 0891019)	0891067

Bluefire	
Description	**Part Number**
Aspire S	
Aspire S 4VOIP/Bluefire IX2010 Bundle (IP Card and Router)	0892033
Aspire S 4VOIP/Bluefire IX1035 Bundle (IP Card and Router)	0892034
Aspire S/Bluefire IX1035-T 4 Port Bundle (IP Card and Router with T1 and no VPN support)	0892037
Aspire IP Telephone/Bluefire IX2010 Bundle (Contains 2 0890065 IPhones and Router)	0892029
Aspire IP Telephone/Bluefire IX1035 Bundle (Contains 2 0890065 IPhones and Router)	0892030
Aspire M/L/XL	
Aspire 4VOIP/Bluefire IX2010 Bundle (IP Card and Router)	0892025
Aspire 16VOIP/Bluefire IX2010 Bundle (IP Card and Router) - Requires PALA (0891039) or NTCPU-B1 (0891038)	0892026
Aspire 4VOIP/Bluefire IX1035 Bundle (IP Card and Router)	0892027
Aspire 16VOIP/Bluefire IX1035 Bundle (IP Card and Router) - Requires PALA (0891039) or NTCPU-B1 (0891038)	0892028
Aspire IP Telephone/Bluefire IX2010 Bundle (Contains 2 0890065 IPhones and Router)	0892029
Aspire IP Telephone/Bluefire IX1035 Bundle (Contains 2 0890065 IPhones and Router)	0892030
Aspire/Bluefire IX1035 IAD 16 Port Bundle (16 Port IP Card and Router with T1) - Requires PALA (0891039) or NTCPU-B1	0892035
Aspire/Bluefire IX1035 IAD 4 Port Bundle (IP Card and Router with T1)	0892036

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Desi Labels	
Description	**Part Number**
DESI Pre-Print Labels:	
Desi Professional Print - 1-99	780487
Desi Professional Print - 100-199	780488
Desi Professional Print - 200-499	780489
Desi Professional Print - 500-999	780490
Desi Professional Print - 1000+	780491
Analog Telephones:	
Desi Labels for 780020/780021 - Black, Package of 25	780400
Desi Labels for 780020/780021 - Green, Package of 25	780401
Desi Labels for 780020/780021 - Silver, Package of 25	780402
Desi Labels for 780020/780021 - White, Package of 25	780403
Desi Labels for 780025/780026 - Black, Package of 25	780404
Desi Labels for 780025/780026 - Green, Package of 25	780405
Desi Labels for 780025/780026 - Silver, Package of 25	780406
Desi Labels for 780025/780026 - White, Package of 25	780407
Desi Labels for 780034 - Metal Silver, Package of 25	780450
2-Button Telephones:	
Desi Label for 2BTN Tel-Silver (25-pkg), For Laser Jet printers only	0893704
Desi Label for 2BTN Tel-Solid Silver (25-pkg), For Ink Jet and Laser Jet printers - solid silver	0893764
Desi Label-2BTN Tel-Black (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893724
Desi Label for 2BTN Tel-White (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893734
Desi Label for 2BTN Tel-Silver (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893744
Desi Label for 2BTN Tel-Silver (25-pkg), No labels on fixed feature keys, used for other languages	0893774
Desi Labels for 2BTN Tel (25-pkg) - Blank, Used with Desi Pre-Print Software for printing large pictures	0893784
4-Button Telephone (P/N 0890072):	
Desi Label for IP1NA-4TIXH Tel (BK)-Black (25-pkg)	780446
Desi Label for IP1NA-4TIXH Tel (MG)-Green (25-pkg)	780447
Desi Label for IP1NA-4TIXH Tel (MS)-Gray (25-pkg)	780448
Desi Label for IP1NA-4TIXH Tel (WH)-White (25-pkg)	780449
22-Button Non-Display Telephones:	
Desi Label for 22BTN Tel-Silver (25-pkg), For Laser Jet printers only	0893700
Desi Label for 223BTN Tel-Solid Silver (25-pkg), For Ink Jet and Laser Jet printers - solid silver	0893760
Desi Label for 22BTN Tel-Black (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893720
Desi Label for 22BTN Tel-White (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893730
Desi Label for 22BTN Tel-Silver (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893740
Desi Label for 22BTN Tel-Silver (25-pkg), No labels on fixed feature keys, used for other languages	0893770
Desi Labels for 22BTN Tel (25-pkg) - Blank, Used with Desi Pre-Print Software for printing large pictures	0893780

Desi Labels	
Description	**Part Number**
22-Button Display Telephones:	
Desi Label for 22BTN Display Tel-Silver (25-pkg), For Laser Jet printers only	0893701
Desi Label for 22BTN Display Tel-Solid Silver (25-pkg), or Ink Jet and Laser Jet printers - solid silver	0893761
Desi Label for 22BTN Display Tel-Black (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893721
Desi Label for 22BTN Display Tel-White (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893731
Desi Label for 22BTN Display Tel-Silver (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893741
Desi Label for 22BTN Display Tel-Silver (25-pkg), No labels on fixed feature keys, used for other languages	0893771
Desi Labels for 22BTN Display Tel (25-pkg) - Blank, Used with Desi Pre-Print Software for printing large pictures	0893781
34-Button Display Telephones:	
Desi Label for 34BTN Display Tel-Silver (25-pkg), For Laser Jet printers only	0893702
Desi Label for 34BTN Display Tel-Solid Silver (25-pkg), For Ink Jet and Laser Jet printers - solid silver	0893762
Desi Label for 34BTN Display Tel-Black (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893722
Desi Label for 34BTN Display Tel-White (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893732
Desi Label for 34BTN Display Tel-Silver (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893742
Desi Label for 34BTN Display Tel-Silver (25-pkg), No labels on fixed feature keys, used for other languages	0893772
Desi Labels for 34BTN Display Tel (25-pkg) - Blank, Used with Desi Pre-Print Software for printing large pictures	0893782
34-Button Super Display Telephones:	
Desi Label for 34BTN Display Tel-Silver (25-pkg), For Laser Jet printers only	0893703
Desi Label for 34BTN Display Tel-Solid Silver (25-pkg), For Ink Jet and Laser Jet printers - solid silver	0893763
Desi Label for 34BTN Display Tel-Black (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893723
Desi Label for 34BTN Display Tel-White (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893733
Desi Label for 34BTN Display Tel-Silver (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893743
Desi Label for 34BTN Super Display Tel-Silver (25-pkg), No labels on fixed feature keys, used for other languages	0893773
Desi Labels for 34BTN Super Display Tel (25-pkg) - Blank, Used with Desi Pre-Print Software for printing large pictures	0893783

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Desi Labels

Description	Part Number
110-Button DSS Consoles:	
Desi Label for 110 DSS-Silver (25-pkg), For Laser Jet printers only	0893705
Desi Label for 110 DSS-Solid Silver (25-pkg), For Ink Jet and Laser Jet printers - solid silver	0893765
Desi Label for 110 DSS-Black (25-pkg), For Ink Jet and Laser Jet printer with light striping	0893725
Desi Label for 110 DSS-White (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893735
Desi Label for 110 DSS-Silver (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893745
Desi Label for 110 DSS-Silver (25-pkg), No labels on fixed feature keys, used for other languages	0893775
Desi Label for 110 DSS-Silver (25-pkg) - Blank, Used with Desi Pre-Print Software for printing large pictures	0893785
24-Button DLS:	
Desi Label for 24 DLS-Silver (25-pkg), For Laser Jet printers only	0893706
Desi Label for 24 DLS-Solid Silver (25-pkg), For Ink Jet and Laser Jet printers - solid silver	0893766
Desi Label for 24 DLS-Black (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893726
Desi Label for 24 DLS-White (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893736
Desi Label for 24 DLS-Silver (25-pkg), For Ink Jet and Laser Jet printers with light striping	0893746
Desi Label for 24 DLS-Silver (25-pkg), No labels on fixed feature keys, used for other languages	0893776
Desi Label for 24 DLS-Silver (25-pkg) - Blank, Used with Desi Pre-Print Software for printing large pictures	0893786

Spare Parts

Description	Part Number
Aspire Handset without cord - White	NSG-170108-001
Aspire Handset without cord - Black	NSG-170108-002
Aspire Faceplate Kit - 22B Non-Display Telephone, Package of 10 with 40 Rivets	0892000
Aspire Faceplate Kit - 22B Display Telephone, Package of 10 with 40 Rivets	0892001
Aspire Faceplate Kit - 34B Display Telephone, Package of 10 with 40 Rivets	0892002
Aspire Faceplate Kit - 34B Super Display Telephone, Package of 10 with 40 Rivets	0892003
Aspire Faceplate Rivets, Package of 25	0892005
Aspire Plastic Wall Mount Plate, Package of 10	0892007
Aspire Wall Mount Hooks-All Telephones, Package of 10	0892008
Aspire 2B DSLT Metal Wall Plates, Package of 5	0892009
Bottom Directory Tray (Dial card)	0893004
Flying Directory Plastic Clip, Package of 10	780525
Flying Directory Card Kit (10 Directory cards for 780525)	770626

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Discontinued Product Service: 900-990-2541
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Emergency Technical Service (After Hours) 203-929-7920
(Excludes discontinued products)

Have any comments, suggestions or corrections for this guide?

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