Release Notes

UNIVERGE SV8100
Family of Products

Main System Features/
Hardware/Mobility/Applications/Terminal

Software Versions R3.0

NEC

1. Overview

The **UNIVERGE SV8100** provides new and enhanced capabilities with the release of Versions R3.0 Software and related hardware. A short description of these capabilities is outlined in the following pages. This release note provides a quick reference of R3.0 and enhancements and may not cover all service and operation conditions. Please refer to the UNIVERGE SV8100 Features and Specifications Manual and the UNIVERGE SV8100 Hardware Manual for additional information.

HARDWARE

2. New 9.5" Expansion Chassis

2.1 Description

The 9.5" chassis (CHS2U B-US chassis) supports 3 blades and in an IP environment, the system capacity of the 9.5" chassis equipped with the Expansion Chassis is the same as that of the 19" chassis (512 IP stations x 200 SIP Trunks). The 9.5" chassis equipped with the Expansion Chassis provides the same number of slots (6) as the 19" chassis. UNIVERGE SV8100 Release 3.0 system software supports combining the 9.5" Base and 9.5" Expansion Chassis to have six universal blade slots for legacy line/trunk blade, and In-skin Application Blades.

Figure 2.1 9.5" Base and Expansion Chassis
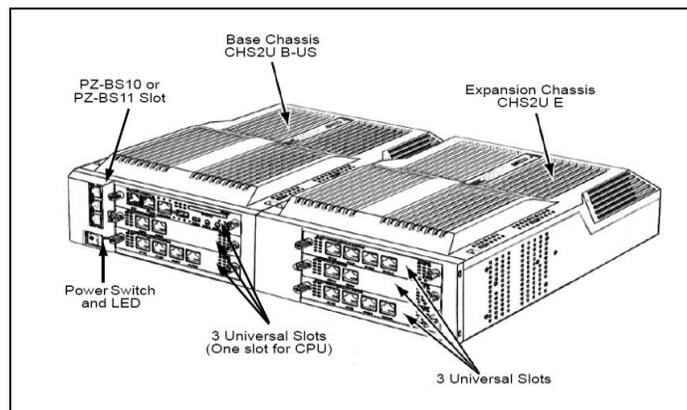
Using two screws, secure the Base and Expansion chassis together with the metal reinforcement bracket provided.

Figure 2.2 9.5" Base and Expansion Chassis Combined
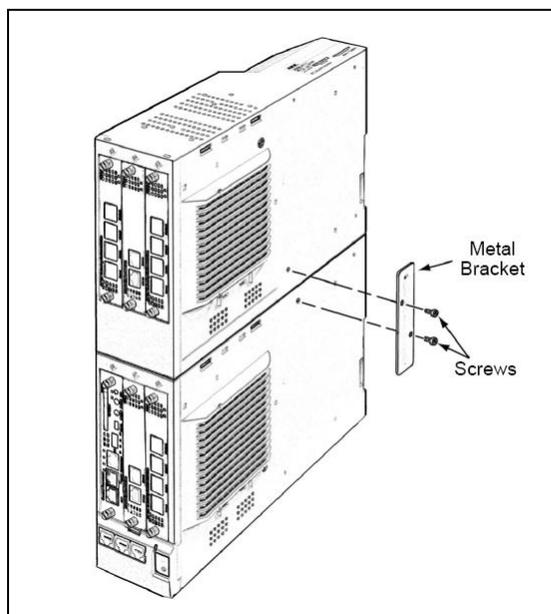
Possible configurations:
- Base + Expansion = 6 slots
- Base + 19" Chassis = 9 slots
- Base + Expansion X 4 = 24 slots
- Base + 3 19" Chassis = 21 slots
- Base + Expansion + 3 19" Chasses = 24 slots

Stock numbers for base chassis and bundled packages will remain the same. The CHS2U B-US will replace the CHS2U GW-US in both the bundle and individual stock number beginning Aug 7, 2009.

2.2 Benefits

The 9.5" Chassis addresses the need for a smaller, more cost-effective, alternative at a remote location in the NetLink environment. The new 9.5" Expansion Chassis protects the customers' initial investment by allowing them to purchase the hardware they need and simply add the 9.5" chassis to meet their business growing needs.

2.3 Conditions

- Not compatible with 9.5" chassis (CHS2U GW-US) purchased before Aug 7, 2009.
- 9.5" is not rack-mountable using a standard telephony rack. Mounting for all configurations starting with a 9.5" chassis must be vertical because the 9.5" chassis does not contain fan units and must rely on vents for air circulation. The Rack Mount Bar set is required to support mounting the 9.5" base and expansion chassis to a rack.

  Mount as depicted in the following diagrams:

Figure 2.3 Attach Rack Mount Brackets to 9.5" Chassis
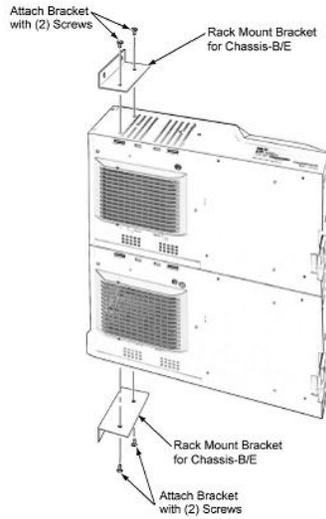
Figure 2.4 Attach Horizontal Brackets to Rack
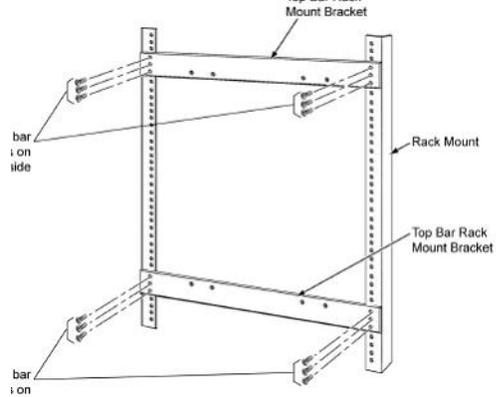
Figure 2.5 Attach (CHS2U B-US) AND (CHS2U E) Chassis to Rack
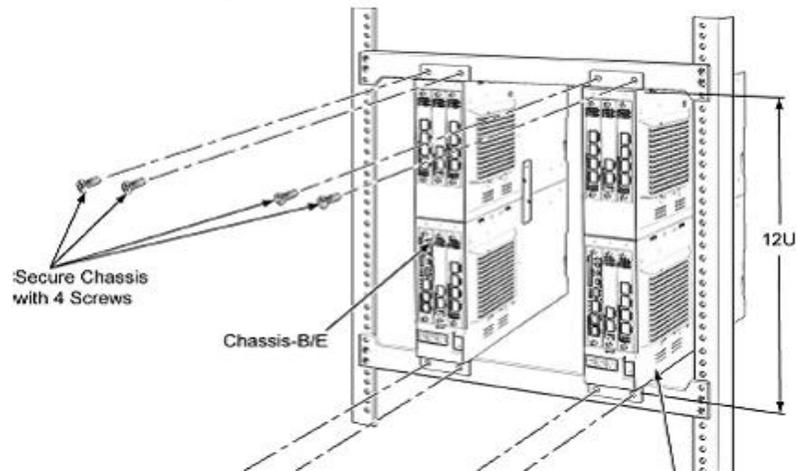
- As a Remote in a NetLink environment, the same conditions apply as a 19" chassis…no SIP trunks can be installed.

2.4 Required Software and Hardware

Version 3.0 Software – available through a free download from the NTAC website (no license required)

MOBILE EXTENSION

3. Call Forward Both-Ring to Voice Mail

3.1 Description

Mobile Extension is an external telephone (preferably a mobile phone) linked to the UNIVERGE SV8100 via a Proxy Port to operate as an internal single line telephone extension. The extension sends DTMF signals to the system allowing access to the system features.
UNIVERGE SV8100 Release 3.0 system software supports
Call Forward-Both Ring to Voice Mail.

The user sets Call Forward – Both Ring to the Mobile Extension. The user then sets the Mobile Extension to forward on either Busy/No Answer or No Answer to the Voice Mail Pilot. Calls not answered at either location will forward to the voice mailbox of the originally called extension. *See example under Conditions.*

REMINDER: Calls will <u>only</u> continue to forward if the Both Ring location (the Mobile Extension) is forwarded (B/NA or NA) to VM and no where else.

3.2 Benefits
With Release 3.0, Mobile Extension provides the ability for users to have one published company number and one voice mailbox as a repository for all messages.

This feature enhancement is an improvement over software versions 2500 and lower, where an extension set to Call Forward-Both Ring would ring both the originating and destination stations until the caller either hung up or the system timer terminated the call, but not route it to any other extension including Voice Mail.

For customers with a few mobile users and a limited budget, Mobile Extension is a viable alternative to costly hardware or service plans.

3.3 Conditions

- In the absence of a physical terminal on the system, a Mobile Extension can still be assigned to a user. Calls made directly to the Mobile Extension can be forwarded to the company voice messaging system when the Mobile Extension is set to Call Forward No Answer to Voice Mail.
  - o Requires ISDN or Analog trunks to operate.
  - o Mobile Extension must be assigned a DID number or a call can be forwarded, transferred or have a line DIL to the Mobile Extension
- Intercom calls placed to the Mobile Extension will hunt to Voice Mail.
- Example:
  - o User has set Call Forward No Answer to VM on Mobile Extension.
  - o Incoming Call rings Ext 200 and Mobile Extension simultaneously.
  - o Call Forward timer expires so the call stops ringing Mobile Extension and is answered by the Voice Mail.
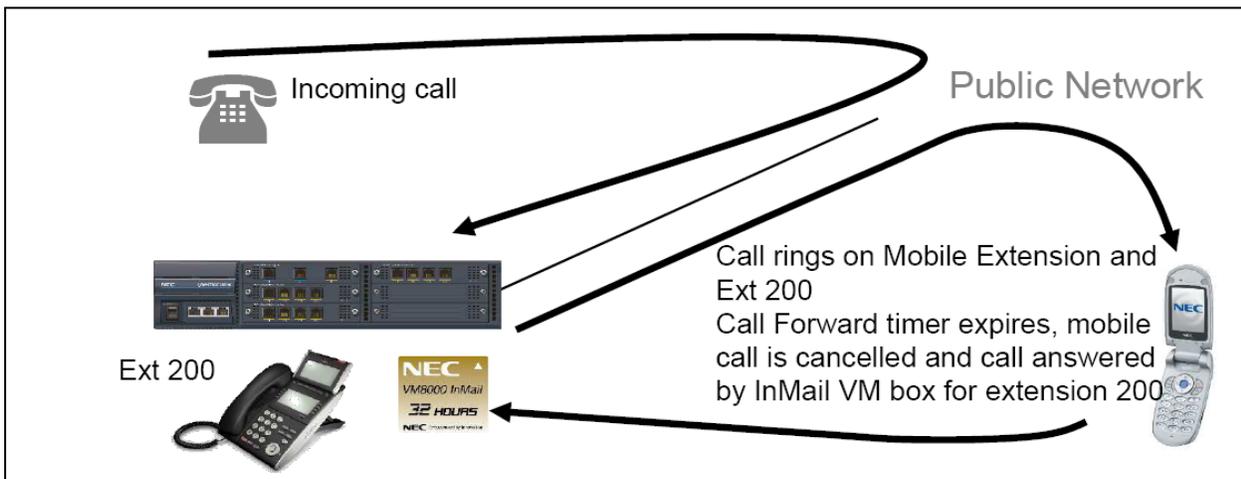
3.4 Required Software and Hardware

Version 3.0 Software – available through a free download from NTAC website (no license required)

4. Caller ID Pass Through for Mobile Extension

4.1 Description

Mobile Extension is an external telephone (preferably a mobile phone) linked to the UNIVERGE SV8100 via a Proxy Port to operate as an internal single line telephone extension. The extension sends DTMF signals to the system allowing access to the system features. UNIVERGE SV8100 Release 3.0 system software supports Caller ID Presented to the Mobile Extension for Type of Call (Direct Internal Call, Direct Trunk Call with CID and Transferred Trunk Call with CID).

Incoming CLI passed to outbound call : Set on a per trunk basis
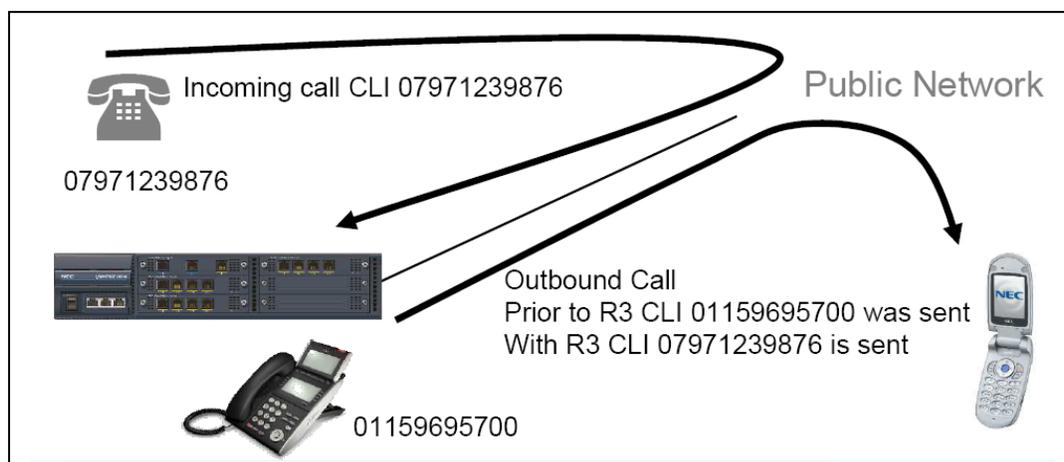
4.2 Benefits

This feature enhancement allows Mobile Extension users to see the Caller ID of the inbound call.

4.3 Conditions

- An ISDN line is required (as these trunks provide answer supervision).
- Direct Internal Call – CPN of the Calling Phone is presented to the Mobile Extension.
- Direct Trunk Call with CID – Caller ID of incoming call is presented to the Mobile Extension.
- Direct Trunk Call without CID – CPN of Mobile Extension is presented to the Mobile Extension.
- Transferred Trunk Call with CID-

- o Transferred **before** inter-digit timeout – Caller ID of incoming call is presented to the Mobile Extension.
- o Transferred **after** inter-digit timeout – CPN of the Transferring Phone is presented to the Mobile Extension.

4.4 Required Software and Hardware

Version 3.0 Software – available through a free download from NTAC website (no license required)

5. Mobile Extension via Analog Trunks

5.1 Description

Mobile Extension is an external telephone (preferably a mobile phone) linked to the UNIVERGE SV8100 via a Proxy Port to operate as an internal single line telephone extension. The extension sends DTMF signals to the system allowing access to the system features. UNIVERGE SV8100 Release 3.0 system software supports Analog trunk for Mobile Extension.

Analog trunks used for integration with the Mobile Extension must be DID or assigned as DILs. Incoming callers access the Mobile Extension by dialing the assigned DID telephone number, by dialing the assigned DIL telephone number or by being transferred by the VRS Auto Attendant. To place a call from the Mobile Extension, the user dials the assigned DID or DIL telephone number.

Analog Trunk   Analog Trunk

2142622000

<Mobile User>

Caller ID: 2142622000

Mobile Extension Port
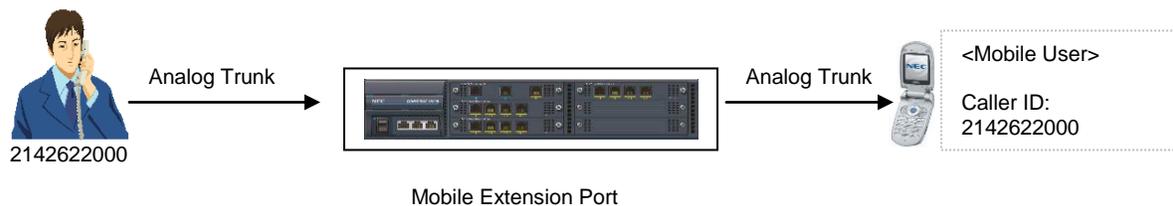
5.2 Benefits

Increases the customer's flexibility and control of operating costs for those users assigned to use the Mobile Extension feature by allowing Analog trunks to be used for Mobile Extension. Prior to UNIVERGE SV8100 release 3.0 system software only supported the use of ISDN (PRI and BRI) trunks.

5.3 Conditions

- In the absence of disconnect supervision, the *0 Hang Up code must be used prior to terminating any call**.**

5.4 Required Software and Hardware

Version 3.0 Software – available through a free download from NTAC website (no license required)

MAIN SYSTEM FEATURES

6. Call Park

6.1 Description

Park places a call in a wait state (called a Park Orbit) so an extension user may pick it up from any terminal on the system permitting a measure of mobility to users without requiring additional hardware. UNIVERGE SV8100 Release 3.0 system software supports parked calls from a virtual extension.

6.2 Benefits

Virtual (secondary) extensions are used to provide a variety of call handling capabilities including receiving calls for other individuals within their group. When the called individual is not at their desk, parking the call and paging that individual is the fastest way for the call to be answered by the intended party. Prior to Release 3.0, calls received on a virtual extension could not be parked without first being transferred to a terminal (primary) extension. The ability to park a call from a Virtual Extension increases productivity by eliminating the extra step.

6.3 Conditions

- An extension user can pick up only a call Parked by a member of their own Park group.
- An extension can have only one Personal Park key.
- A user can display the Caller ID of a call in Park if Caller ID is enabled.

- When a call is parked from a virtual extension, the virtual extension is released.
- Park Group assignment is by the terminal extension, not the virtual extension.
- When a call parked from a virtual extension recalls, it will ring the terminal extension it is programmed on, not the virtual extension key.

6.4 Required Software and Hardware

Version 3.0 Software – available through a free download from NTAC website (no license required)

7. CID Name on Outgoing Calls

7.1 Description

Caller ID is usually a standard feature on ISDN-BRI/PRI provided trunks. Caller ID indication displays the calling party telephone number on the LCD of the multiline terminal for CO incoming calls. This interface provides voice communication path only. UNIVERGE SV8100 Release 3.0 system software supports Calling Party Name sent on outgoing ISDN calls.

7.2 Benefits

Improves customer relationships. Prior to Release 3.0 the calling party identification of calls placed from the SV8100 only listed the number, so for customer oriented businesses like auto dealerships, doctor's offices, banks, etc. reaching their customers was difficult as the recipient often ignored the call.

With this enhancement, Calling Party Names can be set as a system wide parameter or on a per station basis. Note: System wide setting takes priority over station settings.

7.3 Conditions

The UNIVERGE SV8100 supports receiving the name from the Network in supported formats only and can send the Calling Name if allowed by the Service Provider.

Table 7-1 Restrictions for Calling Party Names.

Protocols	Name Delivery Formats
NI-2	Facility Information Elements
4ESS (AT&T Custom)	Not Supported
AT&T5ESS Lucent Custom	Facility Information Element
DMS-100 (Custom) ✶	Display Information Element ✶
DMS-100 (National; ISDN) ✶✶	Facility Information Element ✶✶

✶ *Nortel Specification NIX-A211-1*
✶✶ *Nortel Specification NIS-A233-1*

7.4 Required Software and Hardware

Version 3.0 Software – available through a free download from NTAC website (no license required)

8. Caller ID – Flexible Ringing

8.1 Description

The Caller ID – Flexible Ringing feature provides several different options for rejecting / 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 SV8100 can reject the call by playing a VRS message or it can route the call to an alternative extension or incoming ring group.

8.2 Benefits

The ability to reject (block) or reroute incoming calls without CID or with CID information flagged within an organization as undesirable reduces the number of "unsolicited" calls received during a business day that could interrupt employee productivity. By providing the flexibility of rerouting a call without appropriate CID information to a person or group, valid calls can be accepted and processed.

8.3 Conditions

- This feature does not work with incoming trunk calls via networking (from another system). In this case, the reject/routing parameters must be programmed in the system where the trunks are installed. If rerouting is enabled for those trunks, the calls can be routed to an alternative extension located in a remote system.

8.4 Required Software and Hardware

Version 3.0 Software – available through a free download from NTAC website (no license required)

9. All DSP Busy Alarm

9.1 Description

The UNIVERGE SV8100 system logs various errors and reports information about the operation that can be used to determine the cause of a problem. The system can indicate several errors on the multiline telephone display, output to a USB stick on the CD-CP00-US, or be downloaded in PCPro. UNIVERGE SV8100 Release 3.0 system software supports "All DSP Busy Alarm".

The Alarm message format will vary depending on what type of resource is unavailable and whether the system is stand alone, NetLink or CCISoIP. This will be displayed on telephones and included in printed or emailed Alarm Reports. For example, if there are no VOIP resources when making an IP call, "VOIP All DSP Busy" is shown on the DT700/DT300 display – an Alarm Report is generated also indicating "VOIP All DSP Busy".

The All DSP Busy Alarms can be used to indicate peak calling times within an organization, a need to increase resources by upgrading to a higher capacity IPLA or a maintenance problem.

Alarm types:

Parameters	Description
STA	DSP for IP Station Calls Were All Busy
TRK	DSP for Trunk Calls Were All Busy, includes SIP and CCISoIP Trunks
NET	DSP for CCISoIP Networking Calls Were All Busy
LNK	SIP for NetLink Calls Were All Busy

9.2 Benefits

The addition of All DSP Busy Alarms reports aid the customer in making informed business decisions regarding their DSP resources.

For example: If the alarms are due to an insufficient number of resources being installed, the decision may be to increase the capacity of the IPLA. If the alarms are due to uneven outbound or internodal calling patterns that peak at certain times during the day, changes could be implemented within the organization to stagger the calls reducing the peaks thereby preventing alarm conditions. If the alarms persist and there are sufficient DSP resources to support the offered calling traffic, it could indicate a maintenance issue that needs to be addressed.

9.3 Conditions

- Alarm Reports and System Information Reports can be output to a USB stick on the CD-CP00-US.
- Up to 12 System Alarm times can be scheduled to print on a Monthly, Daily, and Hourly time frame. The report indicates both Major and Minor Alarms.
- System Information Reports cannot be set to output at a scheduled time.
- Up to 100 System Alarm Reports can be stored. When the buffer fills, the oldest record is deleted to allow the new record to be saved.
- The alarm cannot be displayed on Standard SIP Phones, Soft Phones or Single Line phones.

- When using IP Phones, the alarm is shown on both terminals involved in that call if they are both on the same system, this includes NetLink systems.
- If an all DSP condition is encountered when making a call across a CCISoIP Network, only the calling stations receives the alarm indication.

9.4 Required Software and Hardware

Version 3.0 Software – available through a free download from NTAC website (no license required)

10. VM8000 InMail Upload/Download Audio

10.1 Description

The InMail Upload Download Audio feature allows the upload of mailbox greetings up to 1MB in size, recorded on a PC or professionally, to any valid subscriber mailbox in the system. It also allows users to listen to, download and/or delete voicemail messages from callers. Access to the InMail/VRS compact flash drive is via the HTML User Pro (Web Pro).

10.2 Benefits

Mailbox subscribers can upload their own prerecord or professionally recorded greetings directly from their PCs via User Pro. Depending on the recording software employed, users can create crisper greetings more easily than using a telephone and dialpad commands.

Voice mail messages can be downloaded as wav files and archived for future reference, emailed as an attachment to a colleague or included as a permanent record in a client file. Any wav player can be utilized to replay the message (such as Windows Media Player).

10.3 Conditions

- Uploading audio files to any type of Call Routing box and Group mailboxes are not supported. This means auto attendant and group mailbox greetings cannot be uploaded or deleted in the User Pro interface.

- It is not possible to upload/download/delete multiple files simultaneously.
- While uploading an audio file via User Pro the greeting is not accessible by telephone.
- This feature is only supported using a LAN connection.
- In order for uploaded greetings to properly play on the VRS InMail CF they must be in the proper format. Audio files not recorded in the proper format may not playback on the VRS/InMail CF.

   The proper format is:
   - Bit Rate 64kbps
   - Sampling Size 8 bits
   - Channel 1 (Mono)
   - Sampling Rate 8 KHz
   - Audio Format CCiTT u-law

10.4 Required Software and Hardware

- Version 3.0 Software – available through a free download from NTAC website (no license required)
- PZ-VM21
- VM8000 InMail CF

11. PC Pro Enhancements

11.1 Description

PC Pro is a Microsoft Windows based application. It allows the technician/system administrator to download a database from the system, make changes, and then upload.

The key enhancement to PC Pro V3.00 is the System Data Conversion that supports the feature to function conversion of the database from an IPK II to that of an SV8100 during migration

11.2 Benefits

Reduces the number of hours required during installation rekeying all of the existing data into the SV8100 to create the new system database.

11.3 Conditions

- The system data that can be converted is limited to the following:

PRG No.	Name
10-23	IP System Interconnection Setup
11-01	System Numbering
11-02	Extension Numbering
13-04	Abbreviated Dial Number and Name
14-01	Trunk Basic Data Setup
14-02	Analogue Trunk Data Setup
14-05	Trunk Group
14-06	Trunk Group Routing
15-01	Extension Basic Data Setup
15-07	Programmable Function Key
21-02	Trunk Group Routing for Extensions
22-02	Incoming Service Type Setup
22-04	Incoming Ring Group Setup
22-05	IRG Assignment for Normal Ring Trunk
22-07	DIL Assignment
22-11	Dial-In Conversion Table Data Setup
26-01	Automatic Route Selection Service
26-02	Dial analysis Table for ARS/LCR
26-03	ARS Dial Treatments
26-04	ARS Class of Service
26-11	Transit Network ID Table
26-12	Network Specified Parameter Table for ARS/F-Route
40-02	Mail Box Setup
44-02	Dial analysis Table for F-Route Access
44-05	F-Route Table
44-06	Additional Dial Table

11.4 Required Software and Hardware

- Version 3.0 Software – available through a free download from NTAC website (no license required)
- PC Pro V3.00 or higher - available through a free download from NTAC website (no license required)

12. Automatic Overflow (VRS)

12.1 Description

If an incoming trunk call is answered by the VRS Auto-Attendant and the user does not dial an extension, misdials, or the destination is either busy or doesn't answer, Automatic Overflow redirects the call to a designated Ring Group, voice mail or speed dial bin. Previously, it only routed callers to a designated Ring Group or voice mail.

12.2 Benefits

The ability to redirect to a speed dial bin provides access to either internal extensions or external telephone numbers widening the scope of how and where your customer's callers are answered.

12.3 Conditions

- VRS record time is fixed at two minutes and cannot be changed.
- The Automated Attendant (VRS) can answer up to 16 calls simultaneously.
- The maximum number of VRS ports is 16 when the PZ-ME50-US is installed on the CP00. If the PZ-ME50-US is not installed, the maximum number of VRS ports is eight.

12.4 Required Software and Hardware

Version 3.0 Software – available through a free download from NTAC website (no license required)

13. NAPT (Network Address Port Translation)

13.1 Description

This function is used to establish SIP communication with the main device and RTP communication among SIP terminals through the NAT on an IP network.

- Supports Static NAT and Dynamic NAT
  Static NAT is used to convert an address (a WAN-side address for the NAT router).
  Dynamic NAT is used to automatically acquire the WAN-side global address for the NAT router and convert it.
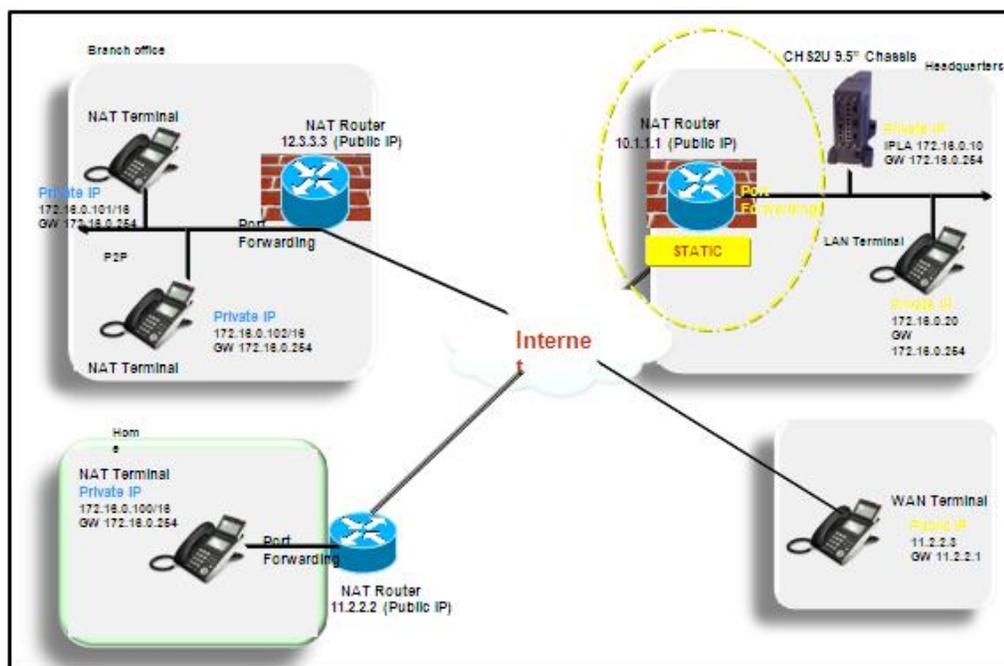
13.2 Benefits

DT700 series terminals can be installed at remote locations without requiring a VPN to access the system. Road warriors can call from anywhere more easily.

Additional benefits include:
- The CPU uses one slot - no extra real-estate is required to implement NAPT.
- NAPT supports all 512 terminals.
- Only one "V3000 Enhancement License" is required for all 512 terminals.
- The number of Public IP addresses required is reduced.
- The SV8100 system remains a Private IP address.
- Encryption is supported.
- The NAT router acts like a firewall between the Public and Private IP addresses providing security to the SV8100.

13.3 Conditions

- "License Exceeded" will display on the DT700 terminal when trying to register a NAT phone, if the feature is not licensed.
- IP terminals can be connected via NAT router or WAN (direct connection).

- The NAT router on the SV8100 side must have a static WAN IP address.
- *Not all routers will support the NAPT configuration.*
- *Softphones (SP310) are not supported.*

13.4 Required Software and Hardware

- Version 3.0 Software – available through a free download from the NTAC website (no license required)
- Feature license on the CD-CP00-US
- Terminals using NAPT must be at firmware V93.0.5.0 or above

TERMINALS

14. DESI-Less Pop Up

14.1 Description

The system can be programmed to blink the page number of a DT300/DT700 DESI-less terminal when it receives an incoming call, or switch to the page the incoming call is on. Furthermore, a default page can be defined for the DESI-less terminal to change to when it goes idle or when it has answered a call.

14.2 Benefits

DESI-Less Pop Up is an innovative way for customers who handle a high call volume and need to quickly answer incoming without scrolling through multiple pages. Manually / Automatically switch to the page that has the activity. Indicates the events on the background DESI less pages.
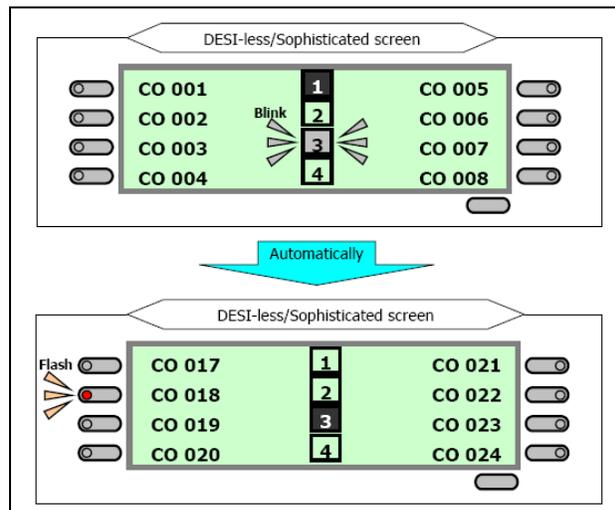
14.3 Conditions

- DT300/DT700 terminals installed in a SV8100 with the IPK/IPK II Migration system do not support the DESI-less page switching and blinking.
- DESI-less screen page switching only applies to idle terminals. If a terminal is not idle, the screen will not switch if another call comes in until the phone goes idle.
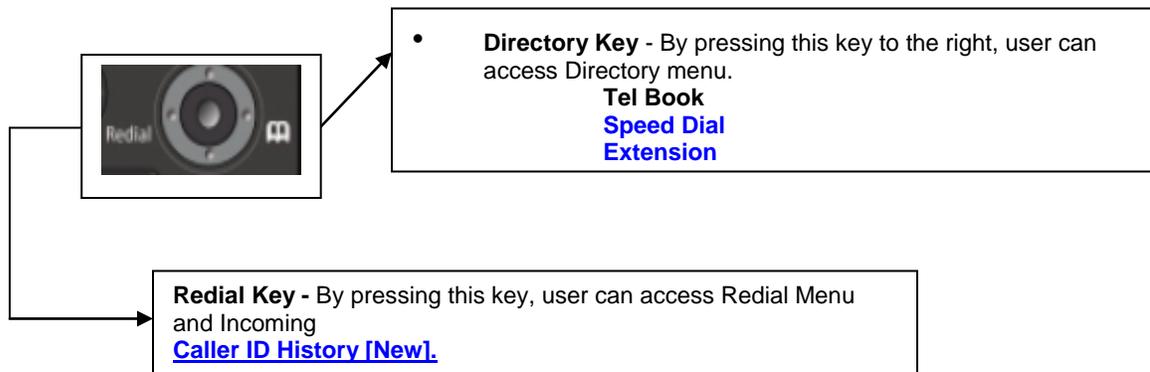
14.4 Required Software and Hardware

Version 3.0 Software – available through a free download from NTAC website (no license required)

15. Caller ID – Cursor Key Operation

15.1 Description

By pressing the Left Cursor Key twice (on equipped terminals) displays the Incoming Call History.

- **Directory Key** - By pressing this key to the right, user can access Directory menu.
  Tel Book
  Speed Dial
  Extension

Redial Key - By pressing this key, user can access Redial Menu and Incoming
Caller ID History [New].

15.2 Benefits

The improvement to the curser key operation provides increased efficiency to terminal users by reducing the number of keystrokes needed to perform routine call processing activities. The enhanced capability allows faster access to all available directories as well as the last 50 incoming calls in the Call History log .

15.3 Conditions

- Once the LEFT key is pressed, the Right Cursor key can be used to switch between the Redial and Incoming History menus.
- If the terminal is not allowed to have the Dial Preview feature, redial menu and directory menus cannot be accessed.

15.4 Required Software and Hardware

Version 3.0 Software – available through a free download from NTAC website (no   license required)

# 16. DT700 Unicast / Multicast Paging Mode

16.1 Description

Unicast transmission is the sending of information packets to a single network destination. It is a one-to one connection between the client and the server. Unicast uses IP delivery methods such as TCP and UDP, which are session based protocols.

Multicast transmission is the sending of information to a group of destinations simultaneously. It is a true broadcast. The multicast source relies on multicast-enabled routers to forward the packets to all client subnets. Each client does not add additional overhead.

IP Terminals can now receive an Internal Page via a Unicast or Multicast packet based upon system programming. This feature allows IP terminals (DT7XX) to be assigned to Unicast Mode, Multicast Mode, or Auto.

Figure Auto-Mode Example
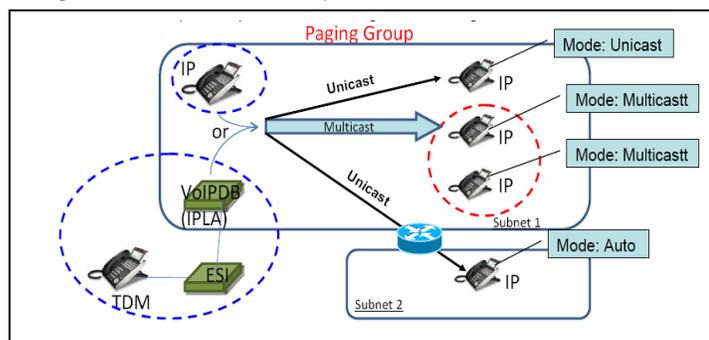
16.2 Benefits

Multicast in corporate environments where all routers are multi-cast-enabled can reduce the amount of bandwidth and resources required to page a group of IP stations.

16.3 Conditions

- Prior to Version 3000 software IP phones utilized Unicast Paging
- When using Unicast mode, there must be an available resource for each IP phone in the page group at the time of the internal page.
- The ability to assign Unicast or Multicast on an IP phone basis, is restricted to internal paging only. Other Multicast features (External MOH, Background Music, Room Monitor) cannot utilize Unicast.
- Number of IP phones per page group is 16.

16.4 Required Software and Hardware

- Version 3.0 Software – available through a free download from NTAC website (no license required)
- DT7XX IP terminal
- CD-CP00-US and CD-(XX)IPLA

17. Multi Logon for Override (phone relocation)

17.1 Description

The same user name and password can be assigned to multiple extensions when using Automatic or Manual Registration. For example if a user has an *IP multiline terminal*, *MH240 handset*, and uses *Desktop Applications* with the Enhancement bundle controlling the IP Multiline, three different ports are used in the system.   The users only have to remember one password.

Prior to Version 3000, each IP port required a unique user name and password.

17.2 Benefits

Users have numerous name and password combinations they need to remember to gain access to the tools they need to perform their assigned tasks. This enhancement reduces that number by allowing them to access multiple desktop tools with the same user name and password.

17.3 Conditions

- When using the Multiple Logon, the same Personal ID index can be assigned to an ITL/Softphone, a CTI (Desktop), and an MH240 terminal type.

17.4 Required Software and Hardware

Version 3.0 Software – available through a free download from the NTAC website (no license required)

18. Registration Override

18.1 Description

When Manual mode is used, Registration Override can be used. Registration Override allows a user to login at one phone, and later login at another phone and keep the same extension number.

Registration Override feature is enhanced to allow *IP Multiline Stations* (SIP) using 1st Party CTI application or 3rd Party CTI application to be overridden giving users access to their IP telephone from any location.

18.2 Benefits

By utilizing the override login function, users have the flexibility of logging into their IP Station in the office then log in remotely from their home office without having to remember to log out of their office terminal before leaving nor login with a separate extension number.

There are several advantages to sharing an extension number between two devices, including but not limited to:
- One voice mailbox
- One History log
- One MW indication
- One published extension number
- One set of login credentials

It's a simple operation…The user logs in at work to use the office terminal, when they get home or are on the road they login their home office terminal or Soft Phone using the same credentials. When this happens the office terminal is logged out automatically and the home office terminal or soft phone is logged in in its place.

18.3 Conditions

- An SIP multiline terminal can override another SIP multiline terminal or a Softphone.
- A Softphone can override another Softphone or a~~n~~ SIP multiline terminal.
- Override does not support SIP multiline terminal with DSS console or Softphone with DSS Console.
- The overriding terminal must be of the same type and have the same number of line keys as the terminal to be overridden.
- Each IP/SIP softphone or terminal must be licensed on the system.

18.4 Required Software and Hardware

Version 3.0 Software – available through a free download from NTAC website (no license required)

19. Software and Hardware

Unit Name	Description	Part
CHS2U B-US	Base Chassis • 9.5" chassis • (670529) AC cord • Installed Blank Slot Covers • Wall and Stand Mounts	670067
CHS2U E	Expansion Chassis • 9.5" chassis • Installed Blank Slot Covers • Wall and Stand Mounts	670068
IP3-RACK MOUNT BAR SET	Rack Mount Set for CHS2U B-US and CHS2U E-US • (4) Chassis Brackets (uses 2 per Chassis) • (2) Horizontal Bar Rack Mounts)	0910008
Release 3.0 Feature License		
LK-SYS-V3000 Enhancement LIC	Release 3.0 Feature License supports : • CPU NAT • KCCIS IP TO IPK/IPKII (future) Requires *Version 3.0 system software* license (available at NTAC download)	670711
InMail Email Licenses		
LK-SYS-VM-EMAIL-LIC	(Includes 5 InMail VM Email clients) Requires : • (670525) PZ-ME50 – US • (670103) PZ-VM21 • (670831) AKS InMail 512M App CF • LKS – VM-InMail Client LIC	670931
LK-VM-EMAIL-CLIENT1 LIC		670926
LK-VM-EMAIL-CLIENT4 LIC		670927
LK-VM-EMAIL-CLIENT8 LIC	Addl InMail VM Email Clients	670928
LK-VM-EMAIL-CLIENT16 LIC		670929
LK-VM-EMAIL-CLIENT32 LIC		670930

19. Technical Documentation

Technical documentation available for download on the Information Portal:

SV8100:

SV8100 / SV8300 Hardware Manual
SV8100 General Description
SV8100 Features and Specifications

VM8000:

SV8100 VM8000 InMail System Guide