# Release Notes

# *UNIVERGE*® SV8300

## R5.0 Software Release

# 1. Overview

**NEC Corporation of America** and the ***UNIVERGE®*** **SV8300** continues to provide new and enhanced features with the release of 8300 R5.0 series software. R5.0 revision software includes new and enhanced features that help to improve, simplify, and increase the functionality of the SV8300.

R5.0 new and enhancement features:

- Mobility Access Enhancements
- Suite Room Service
- PMS Enhancements
- Peer to Peer Standard SIP Station
- Video Phone Polycom VVX1500
- Standard SIP Enhancement - REFER Method
- T.38 FAX Relay on SIP Trunk
- Override Enhancements
- Message Waiting Lamp with Call History
- Separate CID For Outgoing Call on My-line and Sub-line
- 32 Party Conference Enhancements
- Automated Attendant & DISA with SIP Trunk
- SP350 Data Conference with SV8500
- PC Pro Enhancement

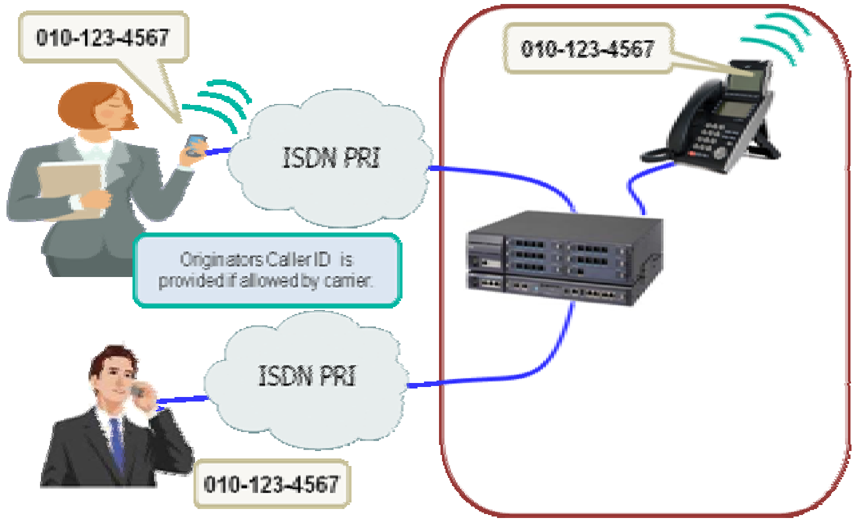
# 2. New and Enhanced Features to R5.0

## 2.1 Mobility Access Enhancements

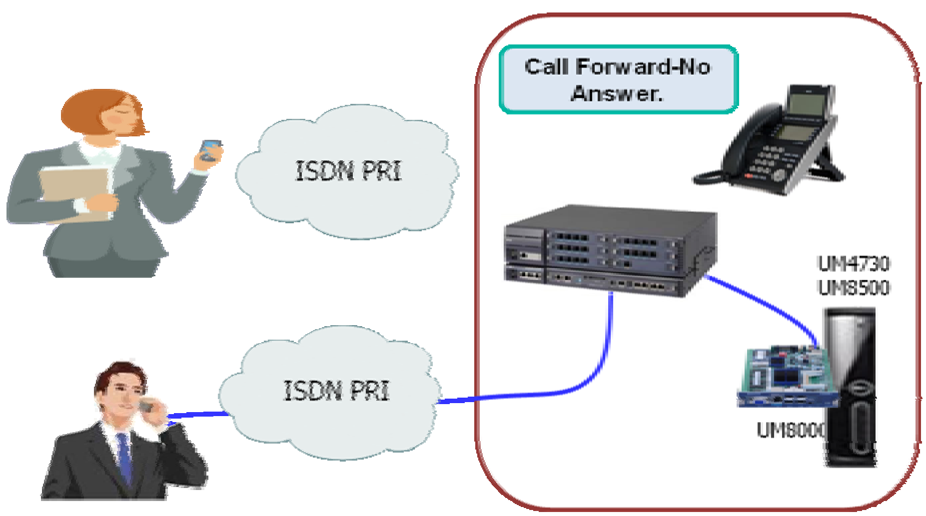
With R5.0 version software the SV8300 continues to provide enhancements to service features for Mobility Access.

**Dual Ringing (Twinning):** Both mobile phone and station ring and when either answers the other stops ringing.
In the existing Mobility Access feature, incoming calls to the station in MA mode are forwarded to the mobile phone immediately without ringing the station in MA mode.
This enhancement provides Dual Ringing (Twining) and allows the SV8300 system to ring both the mobile phone and the station when receiving an incoming call to the station in MA mode, and the user to answer the call with either the mobile phone or the station. This feature allows not only the mobile phone but also the MA station to ring when an incoming call to the MA station is forwarded by MA feature. It also allows the user to answer the call with either the mobile phone or the MA station. When one of the terminals answers the call, the other terminal stops ringing.


**Caller ID Pass Through:** The enhancement provides Caller ID Pass Through to send the original Caller ID from ISDN trunk to the carrier network and display on the mobile phone. Originators Caller ID is provided if allowed by carrier. Enhancements also allow the adding or removal of prefix codes according to the system or carrier service specifications.




**System Voice Mail:** Call Forward-No Answer call will go to system VMS.
Call Forwarding–No Answer for call forwarding in MA mode is added in the Mobility Access features. Call Forwarding–N/A for call forwarding in MA mode is performed only when the system does not receive response signal from the network for a certain period after initiating MA call to the mobile phone. This enhancement allows transferring the call to system voicemail, when a call is terminated to mobile phone (remote MA user) and the mobile phone is busy or no answer.
**Note:** Timer for activating Call Forwarding–N/A for call forwarding in MA mode must be specified in system data.




### 2.1.1 Required Software and Hardware

- R5.0 Version License Software
- MA User License

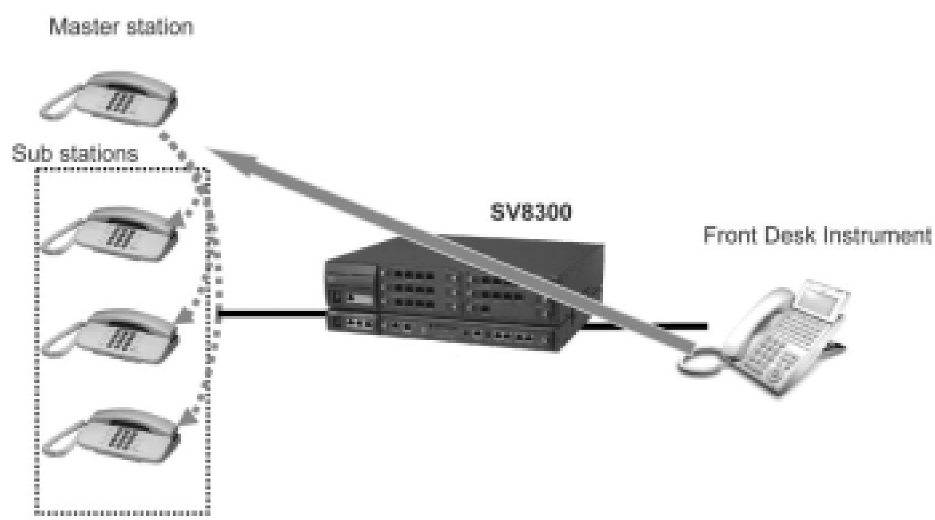
## 2.2 Suite Room Service

With R5.0 version software the SV8300 introduces Suite Room Service. This feature adds to the many Hotel/Motel features and functions available on the SV8300. Up to 128 Suite Rooms are available with four stations servicing one suite room. Each telephone will have its own station number. One station is designated as a master station and the others sub stations.

**Master station:** Single Line Telephones, Digital Multiline Terminals, IP Multiline Terminals, standard SIP Terminal
**Sub station:** Single Line Telephones, Digital Multiline Terminals, IP Multiline Terminals, Soft Phones, PS, standard SIP Terminal

**Calling the suite room station (Suite Room Termination)**
When a call is directed to "master station number" from Front Desk Instrument or other stations, all stations in the suite room ring simultaneously. Any station in the suite room can answer it.

**Originating call from suite room**
From either a master station or a sub station in the suite room, a call can be originated independently. Regardless of the station used in the suite room, master station number is displayed on the LCD on a called party. All the calls are charged to the master station number.




**Additional Suite Service functions:**

– PMS and Call Accounting register the Master Station even if the call is made by Sub station.
– Caller ID displays Master number even if Sub calls
– Message Waiting and Room Cutoff are set to Master and affects Subs
– DND can be set by Master and or Subs
– WU set to Master and rings Subs

#### 2.2.1 Required Software and Hardware

- R5.0 Version License Software
- PMS License

## 2.3 PMS Enhancements

With R5.0 version software the SV8300 continues to provide enhancements to PMS Service features.

- Room Status Codes received from the PMS has been expanded.
  - Using only Front Desk Terminal six user defined codes (3-8) are available.
  - Using PMS/Front Desk Terminal, three user defined codes (0-3) are available.
- Allows simultaneous use of PMS system and Front Desk Instrument. When PMS goes down, allows operating Check In/Check Out function from Hotel/Motel Front Desk Instrument.
- Room status change during PMS system failure can be synchronized, when the PMS system is recovered.
- Room status changed during a PMS failure will be synchronized when the PMS is recovered.

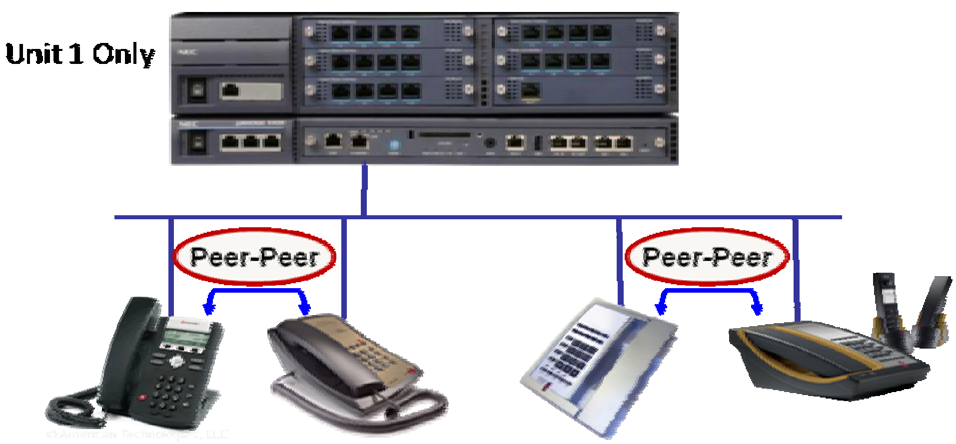
#### 2.3.1 Required Software and Hardware

- R5.0 Version License Software
- PMS License

## 2.4 Peer to Peer Standard SIP Station

With R5.0 version software the SV8300 enhances have been made for Peer-to-Peer calling for Standard SIP stations. With station to station connection between Standard SIP stations within the Main Unit, voice on calls can be connected via Peer-to-Peer without via VoIP-DB. By adding an access code at the head of the station number when originating a call from Standard SIP stations, you can select that the call is via Peer-to-Peer connection or via VoIP-DB connection.

**Note:** Peer-to-Peer connection is available only between IP Standard SIP stations. Peer-to-Peer connection is not available between Standard SIP stations and DT700 Series.

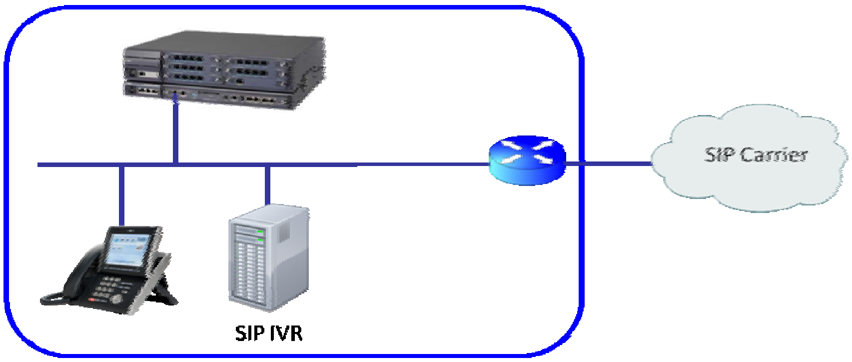



### 2.4.1 Required Software and Hardware

- R5.0 Version License Software
- Standard SIP License

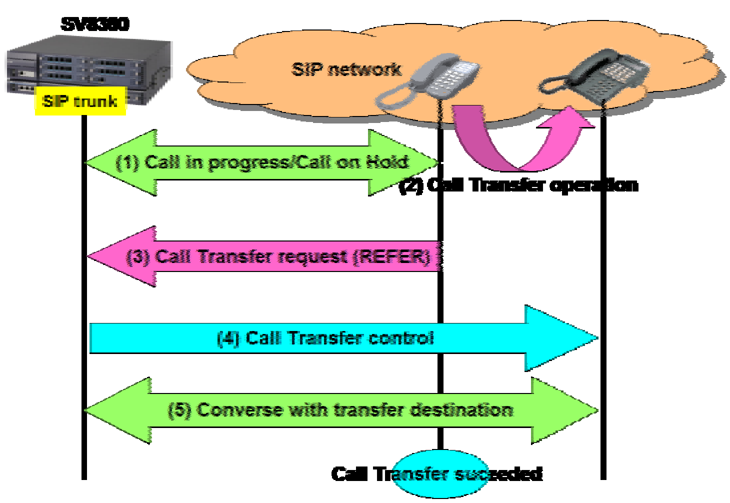
## 2.5 Standard SIP Station Enhancement

With R5.0 version software the SV8300 also added Non-Registration with REFER Method. The enhancement allows the interface of Standard SIP Stations to IVR systems, Voicemail systems, and Gateways. Support of REFER Message (RFC3515) makes it possible to transfer SIP calls from application server to SV8300 according to REFER Message on Standard SIP.




Previously, when the SIP RERER method of transferring calls is performed by SIP network, the SIP trunk rejected the request and Call Transfer is not executed.

This enhancement enables the system to receive REFER message from SIP network. It is provided for when the SIP trunk is a transferee or a transfer destination. When SIP trunk receives SIP REFER method Call Transfer request from SIP network and originates a call for Call Transfer, same trunk number is used for receiving the request and originating the call. Every call to be transferred by receiving REFER message is originated to the SIP network. It is not possible to connect the call to other trunks or stations.




### 2.5.1 Required Software and Hardware

- R5.0 Version License Software
- Standard SIP License
- SIP Trunk System and Channel Licenses

## 2.6 Video Phone Polycom VVX1500 (Revision D)

With R5.0 version software the SV8300 now supports the Video Phone Polycom VVX1500 Rev. D. Video Phone is available between IP Single Line Telephones (SIP) within a same system where the Peer-to-Peer connection is established.


### 2.6.1 Required Software and Hardware

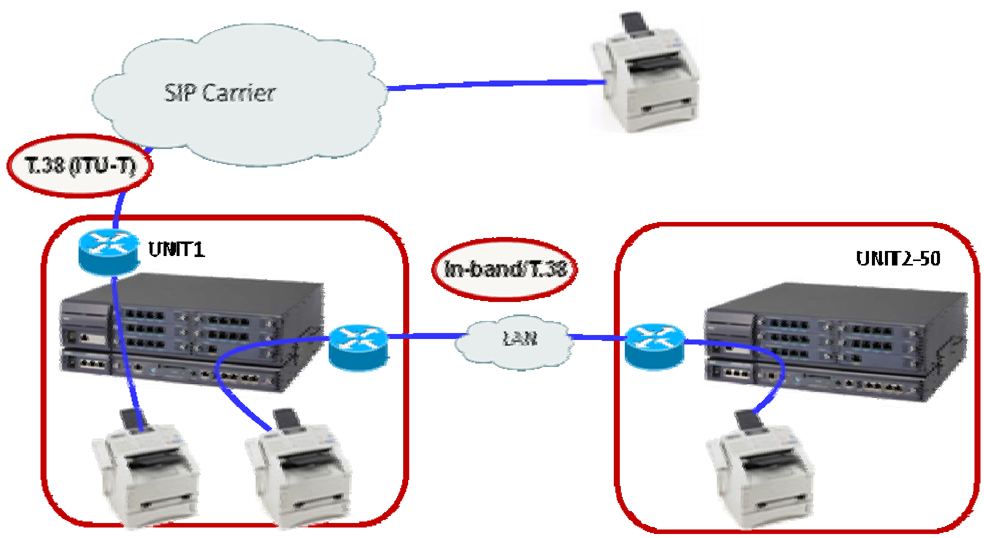
- R5.0 Version License Software
- Standard SIP License

## 2.7 T.38 FAX Relay on SIP Trunk

With R5.0 version software the SV8300 now supports T.38 FAX on SIP Trunks. T.38 FAX communication with SIP carrier (outside) becomes possible and provides stable FAX communication. FAX signals are treated as data packets. T.38 FAX communication can be performed even if delay occurs, as long as the receiver side can receive the packets.

Advantages for using T.38 FAX are that the bandwidth used for FAX communication can be reduced. Also, FAX communication is available regardless of Voice Codec type because FAX communication method is changed to T.38 (UDPTL).

**NOTE:** T.38 FAX testing has been completed with Voxitas and BroadVox SIP carriers. Testing with other SIP carriers is an on going process. Additional information will be posted on the Information Portal under SV8300 Download section.




### 2.7.1 Required Software and Hardware

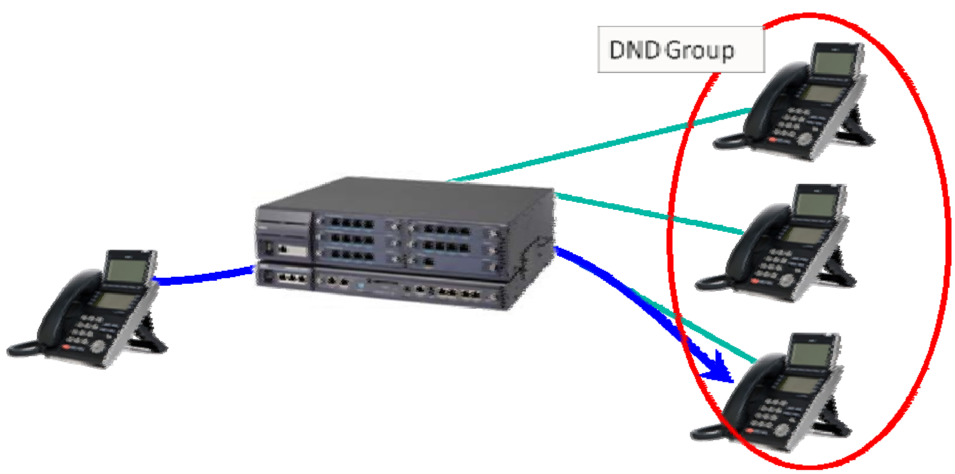
- R5.0 Version License Software
- SIP Trunk System License
- SIP Trunk Channel License

## 2.8 Override Enhancements

With R5.0 version software the SV8300 continues to provide enhancements to service features. Do Not Disturb, DND Group, DND-Hotel/Motel, and Call Forward All Calls can now be overridden by dialing override code and station number.

This feature is effective where called station is in originating system. The services to be overridden by this feature are shown below. Other services work normally.

- DND for individual station
- DND for a group by hotel feature
- DND for a group of stations at pre-programmed times
- Call Forwarding-All Calls (Effective when CM08 - 1014 - 0 is set).
  Call Forwarding to be overridden by this feature is as follows:
  - CM5110 (Forwarding destination of incoming call from other station to a station with DND set in a tenant)
  - CM08-240-0 (Call Forwarding-Busy Line is provided for an incoming call to the station with DND set)




### 2.8.1 Required Software and Hardware

- R5.0 Version License Software

## 2.9 Message Waiting Lamp for Call History

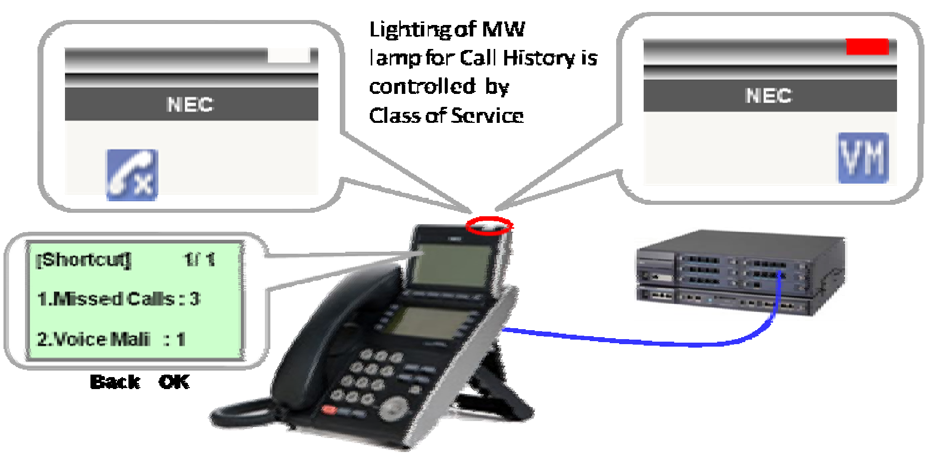
With R5.0 version software the SV8300 continues to provide enhancements to service features. Incoming Call History allows a Multiline Terminal user to display Incoming Call History, which is a list of incoming caller's telephone numbers with date and time stamp. Incoming Call History can save up to the last 50 numbers per Multiline Terminal (10 numbers in default) and up to 32,000 numbers per system. The user can call back from the call history and erase the call history data from system memory.
With the DT300/DT700 terminals, when a missed call is saved whether to light the MW lamp or not can now be selected in system programming via the terminals case of service.

The services that control MW lamp are divided to the following three groups.

- Incoming Call History (CID Call Back), Message Reminder
- UMS8000 Mail, Voice Mail Live Record
- Message Waiting, Message Waiting Console, Message Center Interface (MCI), Open Application Interface (OAI), Voice Mail Live Record-CCIS

This feature is effective to the following lamps.

- – Corner lamp of own station
- – MW lamp of own station (programmable key on Multiline Terminal)
- – My-line key of own station
- – Sub-line key of other station



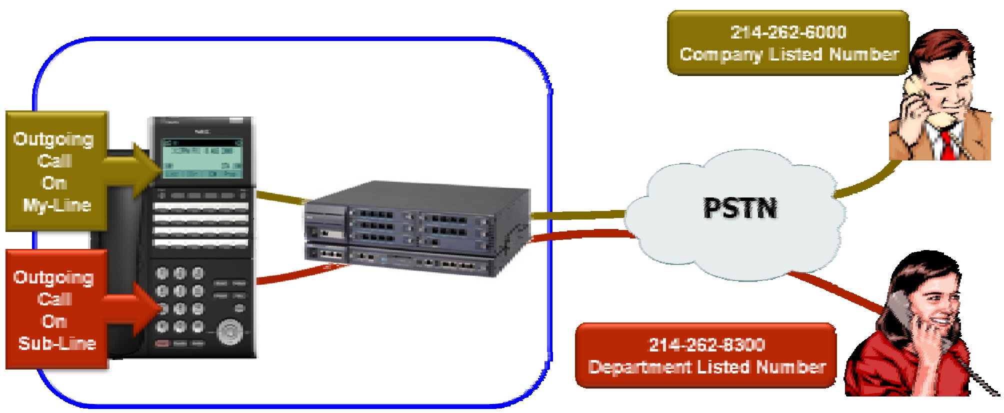

### 2.9.1 Required Software and Hardware

- R5.0 Version License Software

## 2.10 Separate Caller ID for Outgoing Call

This feature is effective when the multiline terminals originate calls using Sub Line to a trunk.  The Sub Line can now have its own Caller ID separate from the My Line Caller ID.  If there is no CID setup for the sub-line or virtual, then the CID of my-line is used.

As an improvement for station outgoing calls to ISDN network, in addition to the existing function that handles two types of Calling Party Number for each My Line, the function that sends the Calling Party Number for each trunk is also provided.
With this enhancement, even from a terminal without Sub Line, not only My Line but also another number, such as Calling Party Number for the department assigned to a trunk can be sent.




### 2.10.1 Required Software and Hardware

- R5.0 Version License Software

## 2.11 32-Party Conference Enhancements

With R5.0 version software the SV8300 continues to provide enhancements to 32-part Conferencing. This enhancement is provided to improve operability of 32-Party Conference built-in CPU.

The three enhancements to 32-Part Conference:

(1) **Mute during 32-party conference**

In the existing feature, mute operation has not available in 32-Party Conferences. This enhancement makes the Mute function available and allows conference participant to mute their line by pressing mute key on multiline terminal.

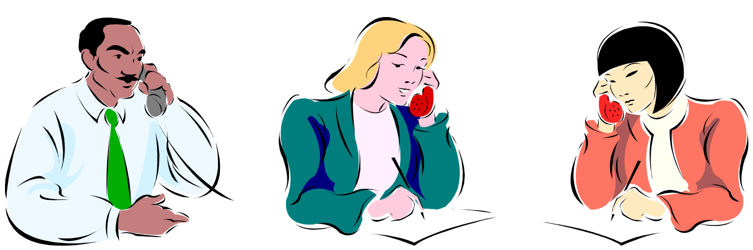
(2) **Conference Connection by Call Transfer**

In the existing feature, only conference participants can add other party to the conference. This enhancement enables a party other than the participants to serve as a participant to execute the operation. This improves convenience especially when adding outside party to the conference.

(3) **Meet-Me conference with password protection**

In the existing feature, any party who knows the pilot station number can enter the conference. There was a risk that the conference is listened by others. This feature allows setting One Time Password and improves security for conference.

- First participant has the capability of setting a one time password after connecting to Conference Trunk.
- All participants use the same password to join the conference.
- The password is automatically cleared when all participants disconnect from the conference.



### 2.11.1 Required Software and Hardware

- R5.0 Version License Software
- LS-CONF-8PORT-LIC (8 conference ports built-in)

## 2.12 Automated Attendant & DISA with SIP Trunk

This enhancement is provided so that Automated Attendant and Direct Inward System Access (DISA) can operate for calls from SIP trunks.

Previously, the SIP Trunk DID number was not available when connected to the Auto Attendant or Direct Inward System Access (DISA). Now the SIP Trunk DID number is provided to Automated Attendant and to Direct Inward System Access (DISA). This can improve convenience for the users.

### 2.12.1 Required Software and Hardware

- R5.0 Version License Software

## 2.13 SP350 Data Conference with SV8500

This feature enables Video/application sharing/Instant Messaging between SP350 in SV8300 and SP350 in SV8500.

Previously, Data conference/Instant Message transmission is available only between SP350s accommodated in SV8300 systems. This feature provides these functions between SP350 in SV8300 and SP350 in the SV8500. Data Conference/Instant Message transmission between SP350 in SV8300 and SP350 in the SV8500 becomes possible. This can improve convenience for the users.

### 2.13.1 Required Software and Hardware

- R5.0 Version License Software
- SP License
- SP-ACD License

## 2.14 PC Pro Enhancements

SV8300 continues to provide enhancements to PC Pro R5 (5.5.0.172)

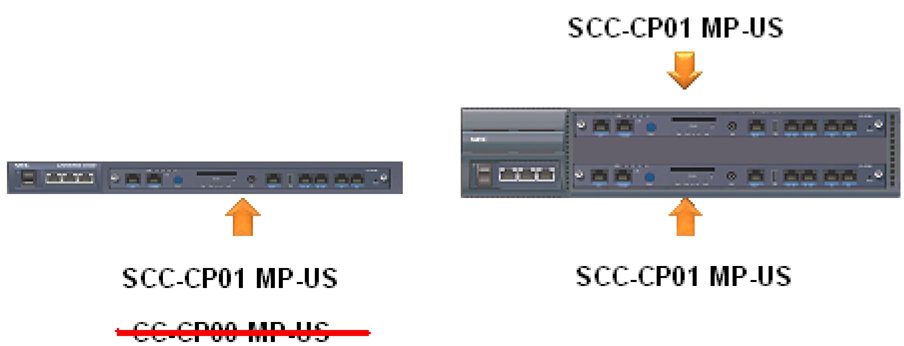
- System Data Change Report
- History of system data setting changes
- Report output in MS Excel format
- Win Server 2008 32/64bit support
- PC Programming Manual

### 2.14.1 Required Software and Hardware

- R5.0 Version License Software
- PC Pro (R5 5.5.0.172)

## 2.15 CP01 for Both Single and Dual CPU Systems

- CP01 will now be used for both Single and Dual CPU systems
- Lower cost migration from Single CPU to Dual CPU system
- Less inventory only one spare CPU stocked for both Single and Dual CPU systems




### 2.15.1 Required Software and Hardware

- All Single and Dual System packages for Main site locations

## Software and Hardware

| Part Number | Description | Comments |
|---|---|---|
| **Software - New and Enhancements** | | |
| **670049** | **LS-SYS-R5-LIC** | **SV8300 System Version License R5.0**<br>Version License loads the following built-in settings in to the main site CPU (UNIT 1):<br>- 108 System Port Licenses<br>- 32 Dterm SIP Licenses<br>- 16 VoIP PAD Channels<br>- 32-Party Conference (8 ports default)<br>- SMDR System Wide License<br>- ISDN System Wide License |
| **670768** | **LS-NW-MA-LIC** | Mobile Access (MA) License - One required for each terminal using the Mobility Access Feature |
| **670769** | **LS-FEA-PMS-LIC** | PMS License - one license per system |
| **670760** | **LS-EXT-STD-SIP-LIC** | Standard SIP License - One required for each Standard SIP terminal |
| **670764** | **LS-TRK-SIP-LIC** | SIP Trunk Channel License - One required per each SIP Trunk channel |
| **670775** | **LS-FEA-SIP-T-LIC** | SIP Trunk - one license per system Enables System SIP Trunk Feature |
| **670859** | **LS-CONF-8PORT-LIC** | 32-Party Conference License - activates additional ports in increments for 8 (8 ports built-in on Version License) |
| **670761** | **LS-EXT-SP-LIC** | SP License - One for each SP350 Softphone |
| **670763** | **LS-EXT-SP-ACD-LIC** | SP-ACD License - One required for each SP350 ACD Softphone |
| **670833** | **AS System PC APP-CD** | SV8300 PC Pro (R5 5.5.0.172) |

| Part Number | Description | Comments |
|---|---|---|
| **Packages / Hardware - New and Enhancements** | | |
| **670173** | **SV8300 BACK-UP CP01 SIP UPGRADE KIT** | SV8300 Back-Up CP01 SIP Upgrade Kit for upgrading existing SIP system from single to dual CPU.<br>Package includes:<br>- SCC-CP01 MP-US (670151)<br>- SCC-CP01 MP-US (670151)<br>- CHS2U-US(D) (670150)<br>- PZ-128IPLA (VoIP DB) (670106)<br>- LA-SYS-DUAL CPU-LIC (670867)<br>- LS-SYS-R5-LIC (670049) |
| **670174** | **SV8300 BACK-UP CP01 SIP UPGRADE KIT** | SV8300 Back-Up CP01 Upgrade Kit for upgrading existing TDM system from single to dual CPU.<br>Package includes:<br>- SCC-CP01 MP-US (670151)<br>- SCC-CP01 MP-US (670151)<br>- CHS2U-US(D) (670150) |

***NOTE:*** *These release notes are provided as a quick reference of R5.0 enhancements and may not cover all service and operation conditions. The UNIVERGE® SV8300 Features and Specifications document should be referenced for detailed information about each feature and enhancement before discussion and implementation.*

## Technical Documentation

| Description | Revision |
|---|---|
| SV8300 Hardware Manual | 4.0 |
| SV8300 Command Manual | 6.0 |
| SV8300 Programming Manual | 6.0 |
| SV8300 Networking Manual | 6.0 |
| SV8300 System Manual | 6.0 |
| SV8300 PC Programming Manual | 5.0 |
| SV8300 System Data Programming | 4.0 |
| SV8300 System Maintenance Manual | 4.0 |
| SV8300 Business/Hotel Feature & Specification | 5.0 |
| SV8300 SMDR-MCI-PMS Specification | 3.0 |
| SV8300 ISDN/Q-SIG Feature & Specification | 5.0 |
| SV8300 CCIS Feature & Specification | 5.0 |
| SV8300 WCS Feature & Specification | 5.0 |
| UM8000 Installation Guide | 2.0 |
| InRouter Configuration Guide | 1.0 |
| PoE Gigabit Switch Configuration Guide | 2.0 |
| SV8300 Parts & Price Book | 7.0 |