



# Product Release Notice

## Synapse 2.2 Software

### SB67070 SIP Gateway

Release Date: April 19, 2012

This document contains the release notes for AT&T Synapse software version 2.2 and the Synapse SIP Gateway. This is an upgrade release with some significant enhancements. Users of previous releases are encouraged to upgrade. The following topics are detailed below.

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## What's New?

The total number of SIP accounts has been increased from a maximum of four to a maximum of ten. Remote Site operations require the use of a SIP account. This change allows additional SIP accounts for both SIP trunking and Remote Site.

## How to Upgrade

Please refer to the [Synapse 2.2 Administrator's Guide](#) for complete system installation steps.

## Version Compatibility

Version 2.2 is compatible with the previous software version 2.0. We recommended upgrading to version 2.2 to benefit from the bug fixes outlined below.

## Recent Changes

SB67030 Deskset and SB67020 Deskset:

| <i>Description</i>   | <i>Reference Number</i> |
|--|-------------------------|
| Corrected an issue on the Voicemail to Email feature where Usernames and Passwords that are not multiples of three characters may cause an authentication failure. | #4900                   |
| Corrected an unreliable operation when switching from Call Appearance to Line Appearance or vice versa which needed a power reset in order to clear the condition. | #4919                   |





Web User Interface (WebUI):

| <i>Description</i>   | <i>Reference Number</i> |
|--|-------------------------|
| Allowed the Voicemail to Email settings to be cleared in order to globally disable the feature.  | #4127                   |
| Fixed an incorrect routing priority when the system has more than 16 interfaces (An interface is defined as a PSTN line, T1 line, or SIP account). | #4164                   |

SB67010 PSTN Gateway:

| <i>Description</i>   | <i>Reference Number</i> |
|--|-------------------------|
| Improved echo cancellation performance on the PSTN Gateway by allowing more aggressive attenuation targets when calibrating the phone lines. | #4905                   |
| Ensured the fine-tuned calibration values for the PSTN Gateway are consistently applied.   | #4966                   |

SB67070 SIP Gateway:

| <i>Description</i>  | <i>Reference Number</i> |
|---|-------------------------|
| Removed the G.711a codec (mainly used in Europe) as an available encoding option for the service. G.711a has been known to cause issues with Broadvox service.  | #4901                   |
| The system will now allow for shorter DID numbers sent from the SIP server, providing they are a minimum of four digits and are within the DID range.   | #4941                   |
| SIP Accounts are now automatically re-registered when a network connection is restored after a disruption.  | #4953                   |
| Optimized the system's audio performance by changing the maximum buffer delay on the WAN and LAN side of the SIP Gateway.   | #4959                   |
| Fixed an error that occurred when using the SIP Gateway at or beyond the maximum 16 channel limit, where the channel resources were not always freed. This error resulted in the SIP Gateway supporting less calls than the advertised maximum. | #4967                   |
| Corrected the misinterpretation of some service discovery protocol messages from the service provider nexVortex.  | #5006                   |
| Incoming calls will now be parsed and routed correctly when multiple SIP accounts start with the same four digits.  | #5007                   |

## Known Problems and Workarounds

Synapse does not currently support the email TLS/SSL encryption required by some email services. Email services that do not require encryption will be compatible with Synapse. For example, one of the more popular free email services is Yahoo. Yahoo's email servers do not need encryption and can be used readily with Synapse's Voicemail to Email feature.

The Outbound SIP Proxy Server field on the SIP Settings webpage should be left blank unless specified by the service provider. In cases where there are multiple proxy servers, the DNS queries may produce incorrect results.





## Manuals

For more information on the features described herein, please reference the Synapse manuals and technical documents located here: <http://smbtelephones.att.com/smb/index.cfm/product-support/manuals/>

## Product Support

For additional questions, please contact the SMB Partner Support team as follows:

US Partners: 1-888-916-2007 (Mon – Fri, 6:00 AM - 5:00 PM Pacific Time)

Canadian Partners: 1-888-883-2474 (Mon – Fri, 6:00 AM - 5:00 PM Pacific Time)

