

# *Milsoft<sup>®</sup> IVR Integration*

*PBX Solutions Using SIP*



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## Introduction

This document describes interoperability between Milsoft® Interactive Voice Response (IVR) and SIP compatible PBX solutions, as well as the underlying data and infrastructure requirements. The Milsoft® IVR complies with Internet Engineering Task Force (IETF) RFC specifications; the RFC number will be referenced where applicable.

## Infrastructure

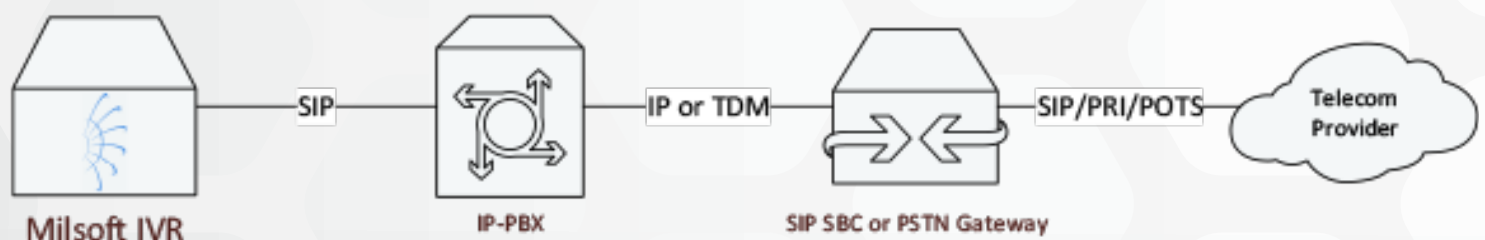
Milsoft® utilizes Internet protocol version 4 (IPv4) for network connectivity. The customer's IP addressing scheme connects to the PBX infrastructure. No public IP addresses are required. Milsoft® IVR is a VMware® compatible virtual machine appliance. Appliance system requirements are available at [www.milsoft.com/resources/hardware-requirements](http://www.milsoft.com/resources/hardware-requirements).

For outbound applications, additional virtual interface boards are recommended. Up to eight boards can be configured per system, with a default of 120 calls per board. Actual call paths are restricted by purchased licensing. For each configured board, a unique SIP integration is required with the PBX system.

Two common topologies for voice integration are shown at a high-level below. Additional topologies can also be supported providing the requirements in this document are met. These topologies do not account for virtualization or network design, as that is outside the scope of this document.

### IP-PBX Topology

In an IP-PBX environment, the Milsoft® IVR integrates directly with the PBX. The PBX then integrates with a Session Border Controller (SBC) or a public switched telephone network (PSTN) gateway to provide telephony access. The PBX is responsible for routing calls between internal phones and the IVR. The SBC/PSTN gateway is responsible for routing calls to the telephone network.



### Traditional PBX Topology

In a traditional PBX environment, the Milsoft® IVR integrates with an SIP SBC, and the SBC integrates to the PBX using traditional TDM technology. The SBC is responsible for routing calls between the PBX and IVR. The PBX is responsible for routing calls to internal phones and the telephone network.



## *Signaling*

Milsoft® uses RFC-3261 SIP signaling to communicate with the customer PBX. Integration can be provided through a shared trunk configuration, or with individual lines per channel. A shared trunk can handle multiple extensions and channels with a single integration configuration. Individual lines require each extension to have a unique integration configuration. The following methods are supported: INVITE, CANCEL, ACK, BYE, OPTIONS, INFO, REFER, NOTIFY. Other SIP methods are unsupported and cannot be required. Only the application/SDP MIME type is supported for message body. Only the SIP URI scheme is supported.

## *Transport*

SIP Transport is compliant with RFC-3261. User Datagram Protocol (UDP) is used for SIP signaling and media transport. Transport Control Protocol (TCP) and SIP Secure over Transport Layer Security (SIPS/TLS) communication are explicitly unsupported. The standard SIP port 5060 is used by default for all communication, and is customizable. For multiple board configurations, the SIP port is incremented by one for each additional board, and must be integrated to the PBX separately. The media traffic port is determined dynamically from the range 49152 to 56352 for each call.

## *Authentication*

Digest authentication for trunk configurations is not required, but is supported. For line-based integration, authentication is required. Realm, extension, username, and password sets must be provided for digest authentication. Extension and username must be unique per trunk or line.

During digest authentication, the FROM contact is digested by the IVR and used for the REFER contact during transfer operations. The IVR requires the FROM contact in the format <sip:{Contact Number}@{PBX ID}>, where {Contact Number} is the calling party from the original call, and {PBX ID} is the PBX IP address or domain name. Additional information inside the angle brackets, such as port identification, is not supported. Any options outside the angle brackets are ignored.

## *Call Identification*

Automatic number identification (ANI) information is required in the SIP INVITE from header for proper caller identification. Dialed number information service (DNIS) information is required in the SIP INVITE to header for proper destination routing.

## Media

### Negotiation

Media is negotiated using the Session Description Protocol (SDP), as defined in RFC-3264 and RFC-4566, as well as the audio profile and definition formats outlined in RFC-3551 and RFC-3555. SDP offers should be present on the initial INVITE request (early-offer). Session renegotiation must be performed through re-INVITE, as the IVR does not support the UPDATE method.

### Payload

The Real-Time Transport Protocol (RTP) is used for all media transmissions, as defined in RFC-3550. The following codecs are supported:

Codec	Bit-rate (Kbps)	Frame Rate (ms)	Frames per Packet	VAD* Support
<b>G.711 <math>\mu</math>-law</b>	64	10, 20, or 30	Fixed at 1	Not Supported
<b>G.711 A-law</b>	64	10, 20, or 30	Fixed at 1	Not Supported
<b>G.723.1</b>	6.3	Fixed at 30	2, or 3	Supported
<b>G.723.1</b>	5.3	Fixed at 30	2, or 3	Supported
<b>G.729 Annex A</b>	8	Fixed at 10	2, 3, or 4	Not Supported
<b>G.729 Annex A with Annex B</b>	8	Fixed at 10	2, 3, or 4	Must be Enabled

### Dual-Tone Multi-Frequency (DTMF)

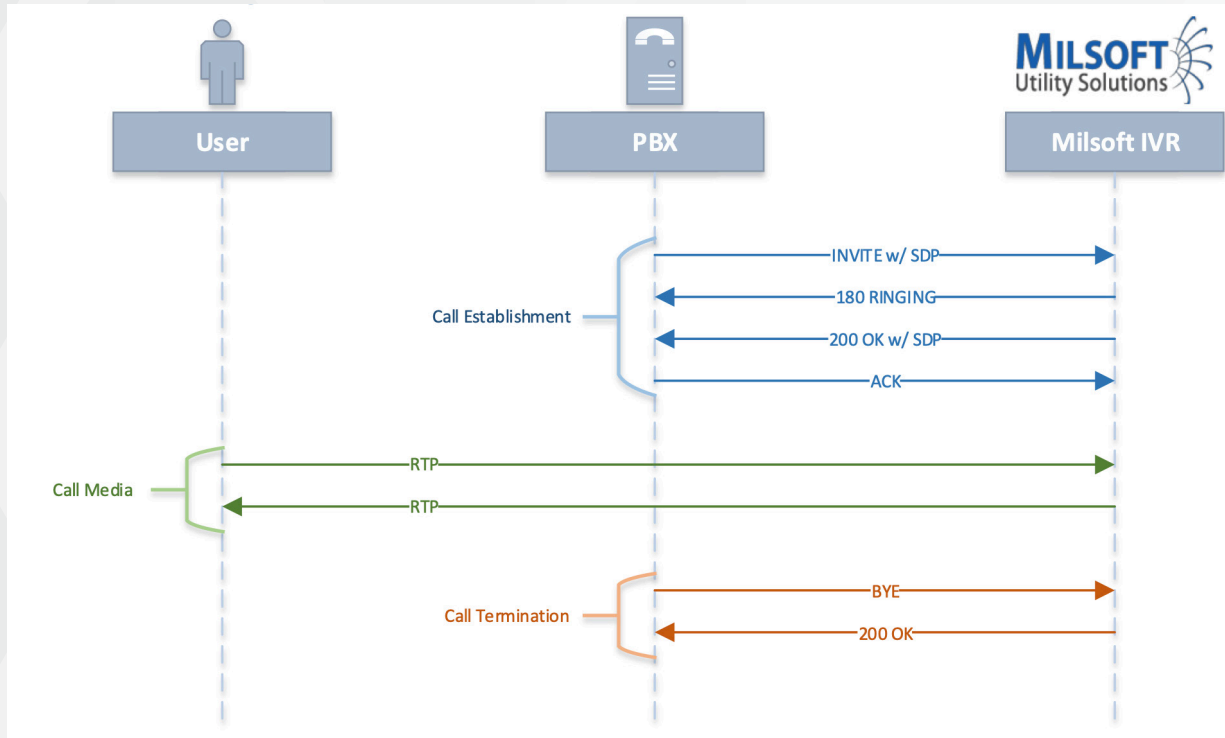
Milsoft® supports RFC-2833 and RFC-4733 out-of-band DTMF only. Audio tones-based DTMF is explicitly unsupported. Any in-band tone-based signals must be disabled or removed prior to reaching the IVR.



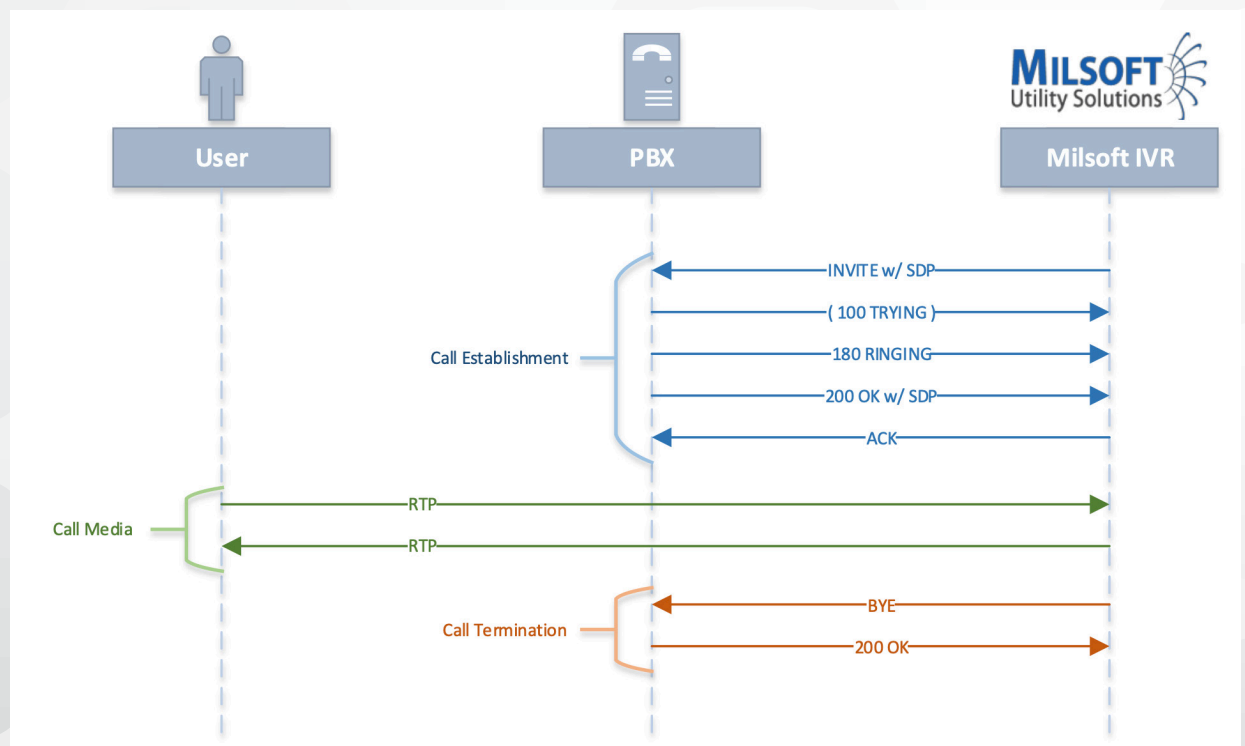
## Call Routing

Call routing between the PBX and IVR follows the basic SIP call flows as outlined in RFC-3665. Transfer examples are given in RFC-5589. Both attended (supervised) and unattended (blind) transfers are supported. Example call flows are provided below for reference. Each example shows interactions involving the Milsoft® IVR only. Additional signaling between the PBX and other endpoints may be required to complete the operations shown, but are outside the scope of this document.

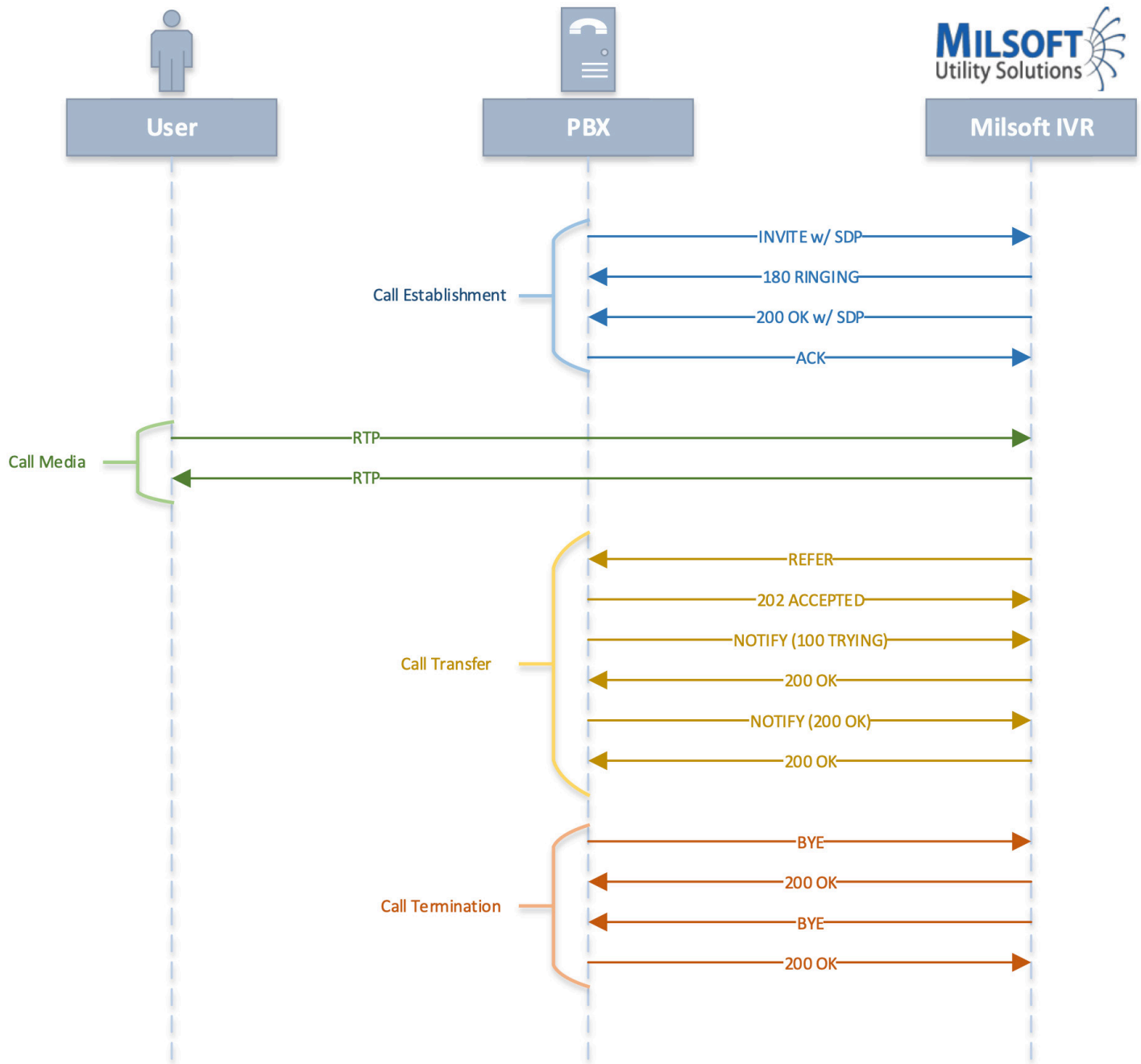
### Inbound Routing



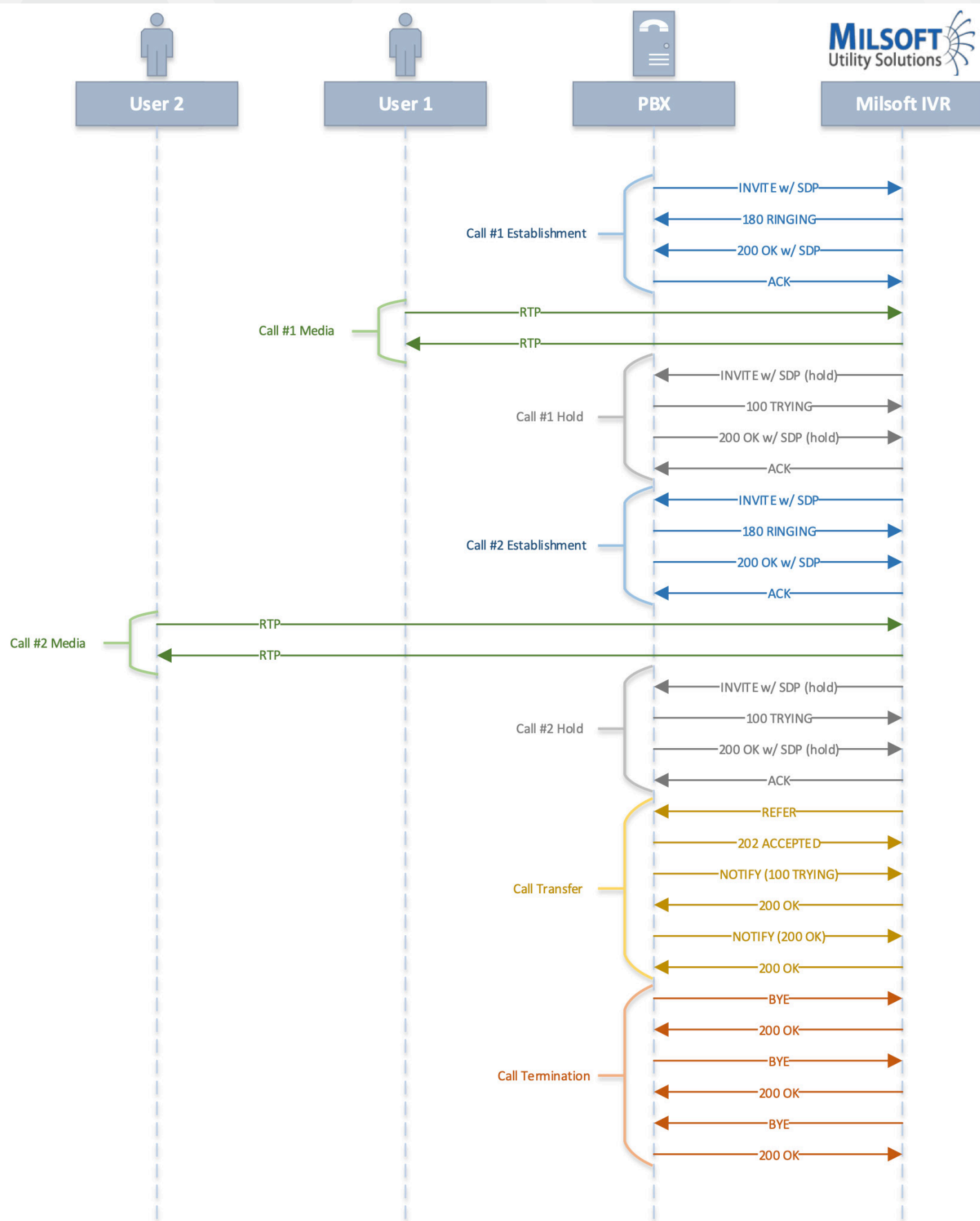
### Outbound Routing



## Unattended (Blind) Transfer



## Attended (Supervised) Transfer





## Appendix A – References

RFC	Title	Link
<b>2833*</b>	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals	<a href="https://tools.ietf.org/html/rfc2833">https://tools.ietf.org/html/rfc2833</a>
<b>3261</b>	SIP: Session Initiation Protocol	<a href="https://tools.ietf.org/html/rfc3261">https://tools.ietf.org/html/rfc3261</a>
<b>3264</b>	An Offer/Answer Model with the Session Description Protocol (SDP)	<a href="https://tools.ietf.org/html/rfc3264">https://tools.ietf.org/html/rfc3264</a>
<b>3550</b>	RTP: A Transport Protocol for Real-Time Applications	<a href="https://tools.ietf.org/html/rfc3550">https://tools.ietf.org/html/rfc3550</a>
<b>3551</b>	RTP Profile for Audio and Video Conferences with Minimal Control	<a href="https://tools.ietf.org/html/rfc3551">https://tools.ietf.org/html/rfc3551</a>
<b>3555</b>	MIME Type Registration of RTP Payload Formats	<a href="https://tools.ietf.org/html/rfc3555">https://tools.ietf.org/html/rfc3555</a>
<b>3665</b>	Session Initiation Protocol (SIP) Basic Call Flow Examples	<a href="https://tools.ietf.org/html/rfc3665">https://tools.ietf.org/html/rfc3665</a>
<b>4566</b>	SDP: Session Description Protocol	<a href="https://tools.ietf.org/html/rfc4566">https://tools.ietf.org/html/rfc4566</a>
<b>4733</b>	RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals	<a href="https://tools.ietf.org/html/rfc4733">https://tools.ietf.org/html/rfc4733</a>
<b>5589</b>	Session Initiation Protocol (SIP) Call Control - Transfer	<a href="https://tools.ietf.org/html/rfc5586">https://tools.ietf.org/html/rfc5586</a>

\*RFC-2833 is obsolete by RFC-4733, but is still used colloquially to refer to the DTMF RTP payload. As such, it is included for reference.

## Appendix B – Glossary

Acronym	Definition
<b>ANI</b>	Automatic Number Identification
<b>DNIS</b>	Dialed Number Information Service
<b>DTMF</b>	Dual-Tone Multi-Frequency
<b>IETF</b>	Internet Engineering Task Force
<b>IVR</b>	Interactive Voice Response
<b>MIME</b>	Multipurpose Internet Mail Extensions
<b>PBX</b>	Private Branch Exchange
<b>PSTN</b>	Public Switched Telephone Network
<b>RFC</b>	Request For Comments
<b>RTP</b>	Real-Time Transport Protocol
<b>SBC</b>	Session Border Controller
<b>SDP</b>	Session Description Protocol
<b>SIP</b>	Session Initiation Protocol
<b>SIPS</b>	Secure Session Initiation Protocol
<b>TCP</b>	Transmission Control Protocol
<b>TDM</b>	Time-Division Multiplexing
<b>TLS</b>	Transport Layer Security
<b>URI</b>	Uniform Resource Identifier
<b>VAD</b>	Voice Activity Detection

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